AD-A156 760 TEST AND EVALUATION OF VIDEO TELECONFERENCING AT 56 KBPS(U) DELTA INFORMATION SYSTEMS INC HORSHAM PA MAR 85 NCS-TIB-85-3 DCA100-83-C-0047 1/2

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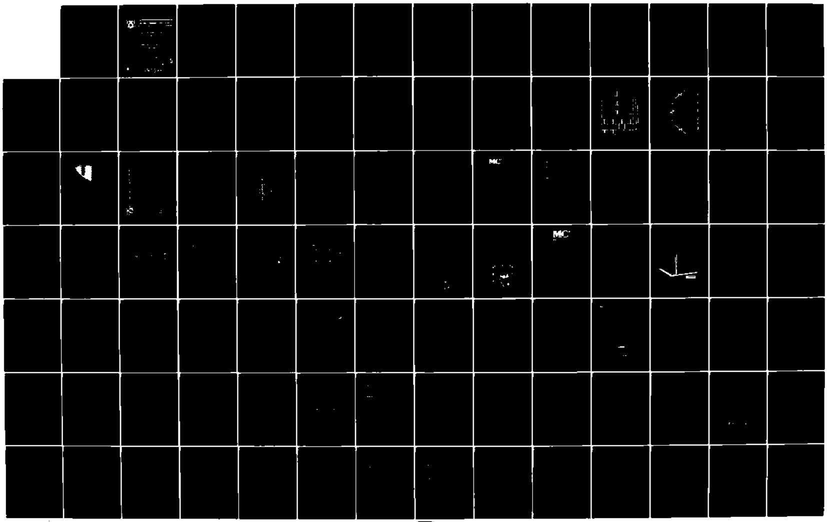

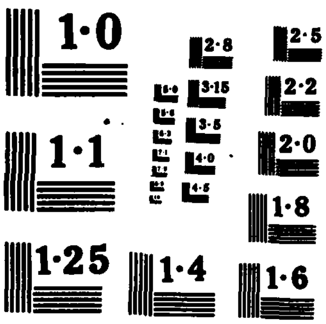
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AD-A156 760

NCS TIB 85-3

NATIONAL COMMUNICATIONS SYSTEM

TECHNICAL INFORMATION BULLETIN
85-3

TEST AND EVALUATION OF VIDEO TELECONFERENCING AT 56 kbps

MARCH 1985

APPROVED FOR PUBLIC RELEASE
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REPORT DOCUMENTATION PAGE		READ INSTRUCTIONS BEFORE COMPLETING FORM
1. REPORT NUMBER NCS-TIB-85-3	2. GOVT ACCESSION NO.	3. RECIPIENT'S CATALOG NUMBER
4. TITLE (and Subtitle) Test and Evaluation of Video Teleconferencing at 56 kbps		5. TYPE OF REPORT & PERIOD COVERED Final Report
		6. PERFORMING ORG. REPORT NUMBER
7. AUTHOR(s)		8. CONTRACT OR GRANT NUMBER(s) DCA100-83-C-0047
9. PERFORMING ORGANIZATION NAME AND ADDRESS Delta Information Systems, Inc. Horsham Business Center, Bldg. 3 300 Welsh Rd., Horsham, PA 19044		10. PROGRAM ELEMENT, PROJECT, TASK AREA & WORK UNIT NUMBERS
11. CONTROLLING OFFICE NAME AND ADDRESS National Communications System Office of Technology & Standards Washington, D.C. 20305-2010		12. REPORT DATE March 1985
		13. NUMBER OF PAGES 123
14. MONITORING AGENCY NAME & ADDRESS(if different from Controlling Office)		15. SECURITY CLASS. (of this report) Unclassified
		15a. DECLASSIFICATION/DOWNGRADING SCHEDULE

16. DISTRIBUTION STATEMENT (of this Report)

Approved for Public Release; Distribution Unlimited.

17. DISTRIBUTION STATEMENT (of the abstract entered in Block 20, if different from Report)

18. SUPPLEMENTARY NOTES

19. KEY WORDS (Continue on reverse side if necessary and identify by block number)

Video teleconferencing, 56 kbps, WIDCOM.

20. ABSTRACT (Continue on reverse side if necessary and identify by block number)

The National Communications System is actively working with the CCITT and ANSI Standards organizations toward the establishment of a standard for video teleconferencing at ISDN communication rates below 1.544 Mbps (e.g. 384 kbps, 64 kbps, 56 kbps). The 56 kbps data race is of particular importance because AT&T has announced a circuit-switched data network operating at this rate which will use the public switched telephone network. The ubiquitous network will provide an excellent resource for future teleconferencing applications. Concurrent with the evolution of the switched 56 kbps over

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data network has been the development of a new generation of motion video codecs operating at 56 kbps. Although the animation distortion of these codecs is more severe than the 1.544 Mbps codecs, it is likely that the image quality will be acceptable for many teleconference applications. The purpose of this project is to determine the acceptability of the video teleconference image quality when operating at 56 kbps. This was accomplished by installing a video teleconference system in an operational environment and evaluating the acceptability of the service.

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NCS TECHNICAL INFORMATION BULLETIN 85-3

TEST AND EVALUATION OF
VIDEO TELECONFERENCING AT 56 kbps

PROJECT OFFICER

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and Standards

APPROVED FOR PUBLICATION:

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Office of NCS Technology
and Standards

FOREWORD

Among the responsibilities assigned to the Office of the Manager, National Communications System, is the management of the Federal Telecommunication Standards Program. Under this program, the NCS, with the assistance of the Federal Telecommunication Standards Committee identifies, develops, and coordinates proposed Federal Standards which either contribute to the interoperability of functionally similar Federal telecommunication systems or to the achievement of a compatible and efficient interface between computer and telecommunication systems. In developing and coordinating these standards, a considerable amount of effort is expended in initiating and pursuing joint standards development efforts with appropriate technical committees of the Electronic Industries Association, the American National Standards Institute, the International Organization for Standardization, and the International Telegraph and Telephone Consultative Committee of the International Telecommuniation Union. This Technical Information Bulletin presents an overview of an effort which is contributing to the development of compatible Federal, national, and international standards in the area of Video Teleconferencing standards. It has been prepared to inform interested Federal activities of the progress of these efforts. Any comments, inputs or statements of requirements which could assist in the advancement of this work are welcome and should be addressed to:

Office of the Manager
National Communications Systems
ATTN: NCS-TS
Washington, DC 20305-2010
(202) 692-2124

TEST AND EVALUATION OF
VIDEO TELECONFERENCING AT 56 KBPS

March, 1985

Final Report

Submitted to:

NATIONAL COMMUNICATIONS SYSTEM

Office of Technology and Standards

Washington, DC 20305

Contracting Agency:

DEFENSE COMMUNICATIONS AGENCY

Contract Number - DCA100-83-C-0047

Task Order 83-C-0047-84-006

D E L T A I N F O R M A T I O N S Y S T E M S, I N C.

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300 Welsh Road

Horsham PA 19044

Table of Contents

Points of Contact

	Organization	Contact
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2.	Federal Emergency Management Agency	Project Manager - G. Clay Hollister System Engineer - Albert Drain
3.	National Security Agency	Systems Integration - John M. Massey - Wayne Gossett
4.	Delta Information Systems, Inc.	President - Richard A. Schaphorst
5.	Widcom, Inc.	Regional Sales Manager - Frank Dobyns
6.	Ilex System, Inc.	Vice President - Ron Burson
7.	Data Products	Sales Representative - Glen Murphy

* References to National Communications System or NCS in this document refer to the Office of the Manager, National Communications System.

1.0 Introduction

This document summarizes work performed by Delta Information Systems, Inc., for the Office of Technology and Standards of the National Communications System, an organization of the U. S. Government, under Task Order number 84-006 of contract number DCA100-83-C-0047. The Office of Technology and Standards, headed by National Communications System Assistant Manager Marshall L. Cain, is responsible for the management of the Federal Telecommunications Standards Program, which develops telecommunication standards whose use is mandatory by all Federal agencies.

The NCS is actively working with the CCITT and ANSI standards organizations toward the establishment of a standard for video teleconferencing at ISDN communication rates below 1.544 Mbps (e.g. 384 kbps, 64 kbps, 56 kbps). The 56 kbps data rate is of particular importance because AT&T has announced a circuit-switched data network operating at this rate which will use the public switched telephone network. The ubiquitous network will provide an excellent resource for future teleconferencing applications.

Concurrent with the evolution of the switched 56 kbps data network has been the development of a new generation of motion video codecs operating at 56 kbps. Although the animation distortion of these codecs is more severe than the 1.544 Mbps codecs, it is likely that the image quality will be acceptable for many teleconference applications. The purpose of this project is

to determine the acceptability of the video teleconference image quality when operating at 56 kbps. This was accomplished by installing a video teleconference system in an operational environment and evaluating the acceptability of the service.

Work on the project was divided into a series of five tasks which are outlined below. This final report is structured to report on the work accomplished on each of these tasks.

Task No.	Title	Section No. in Final Report
1	Selection of Test Equipment	2.0
2	Selection of Test Material	3.0
3	Documentation	4.0
4	Test and Evaluation	5.0
5	Final Report	-
	Conclusions and Recommendations	6.0

2.0 Selection of Terminal Equipment

2.1 Participating Organizations

The work on this project was accomplished by the interaction of several organizations. The role each organization played in the program is summarized below.

- **National Communications System** - Supported the work performed by Delta Information Systems (DIS). DIS performed the following functions:
  - Assisted in the system design
  - Prepared explanatory video tapes
  - Provided several audio components
  - Designed test materials/questionnaire
  - Assisted during the test
  - Prepared final report
- **Federal Emergency Management Agency**
  - System design
  - System installation
  - Test
- **National Security Agency**
  - System design
  - System integration
  - Documentation of system design

2.2 Selected Test System

The test system selected by Delta Information Systems, in conjunction with the organizations mentioned above, to perform this study is presented in this section. Figure 2-1 is a functional block diagram of the test system which illustrates how the terminal subsystems are interconnected to form a complete teleconferencing node.

Table 2-1 is a summary of the terminal equipment selected for the test system. The Crypto Controller from Data Products, the Video Codec from Widcom, and the Vocoder from Ilex are described in detail in Appendices A, B, and C, respectively.

Unfortunately, the Facsimile System from Rapicom could not be made operational for the teleconferencing test. Despite every effort by the FEMA representative, a lack of documentation and product support for the equipment prevented implementation of the facsimile equipment within the test system.

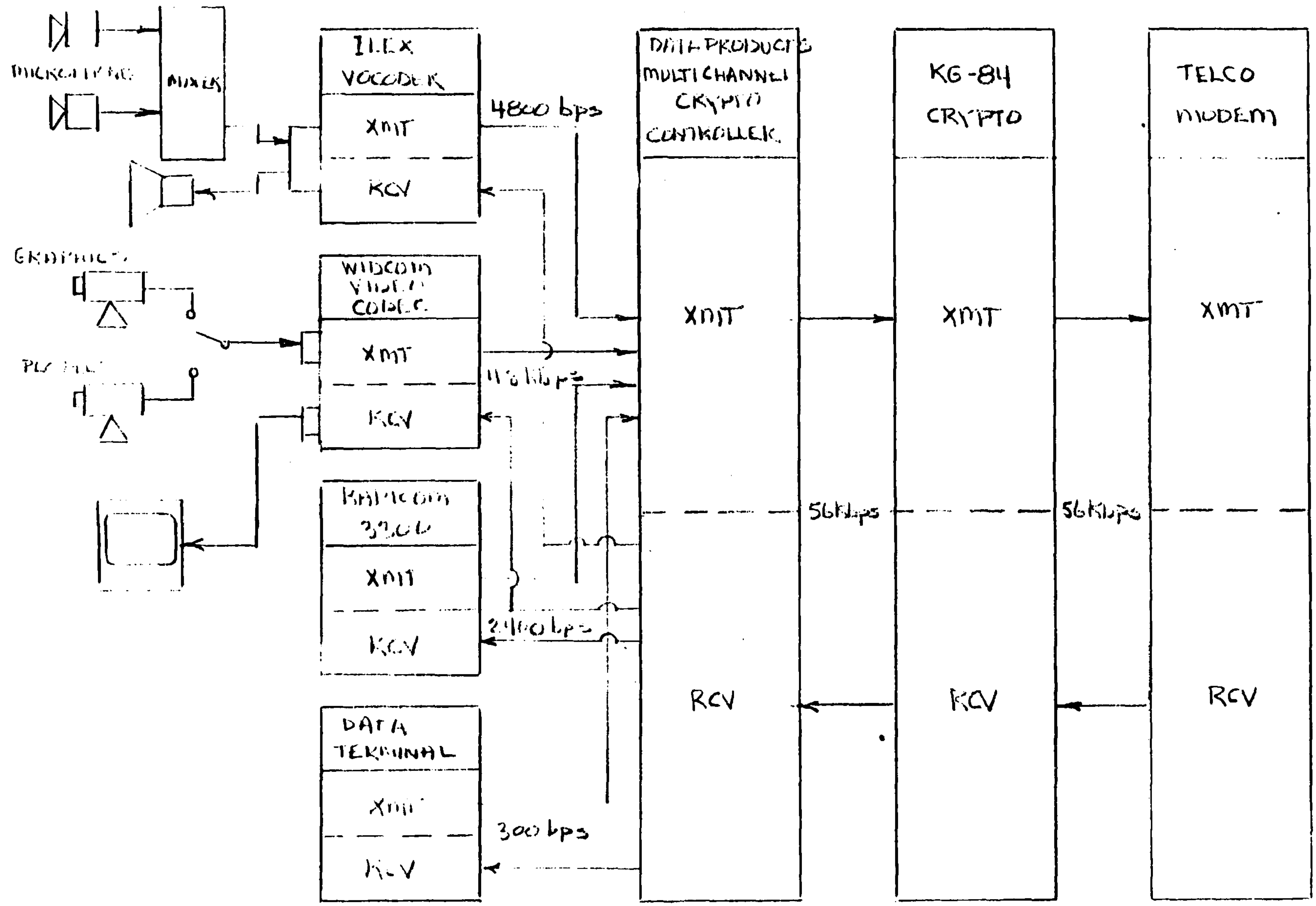


FIGURE 2-1

FUNCTIONAL BLOCK DIAGRAM OF TEST SYSTEM

(Four of these respondents saw the system in "local loopback" mode, without audio.)

3) If you had a requirement for videoconferencing, would this type of system be useful for you?

15	"Yes"
1	"Only in secure mode" (encrypted)

4) What specific agency or department applications do you think the system has (if any)?

Responses:

Point to Point

- HQ to Local Facilities

Point to Multipoint

- HQ Broadcast to Regions
- connect Regions with HQ

Training/Exercises

- Exercises
- Training
- Interagency Field Training Exercises
- Tests and exercises

Conferences/Meetings

- for briefings
- AD/AAD/RD's Meetings
- Regional conferences
- any secure conferences
- IRM Managers Conference
- Staff Mtgs between HQ and SF

Other

- to show disaster scenes
- FEMA Emergency Support Team/FEMA Emergency Response Team
- FEMA Emergency Management Team
- Operations
- use on mobile equipment
- disasters
- many

(The responses were grouped into general categories for convenience.)

5.2 Legibility of Transmitted Graphics

The "Codec" employed in this test provides for the transmission of graphic materials simply by switching the (motion) video input source from one camera to another. Therefore,

5.0 Test and Evaluations

As previously stated, both subjective and objective measurements were taken during this test.

Subjective: Participant Reactions - Individual reactions from those who attended the test were solicited.

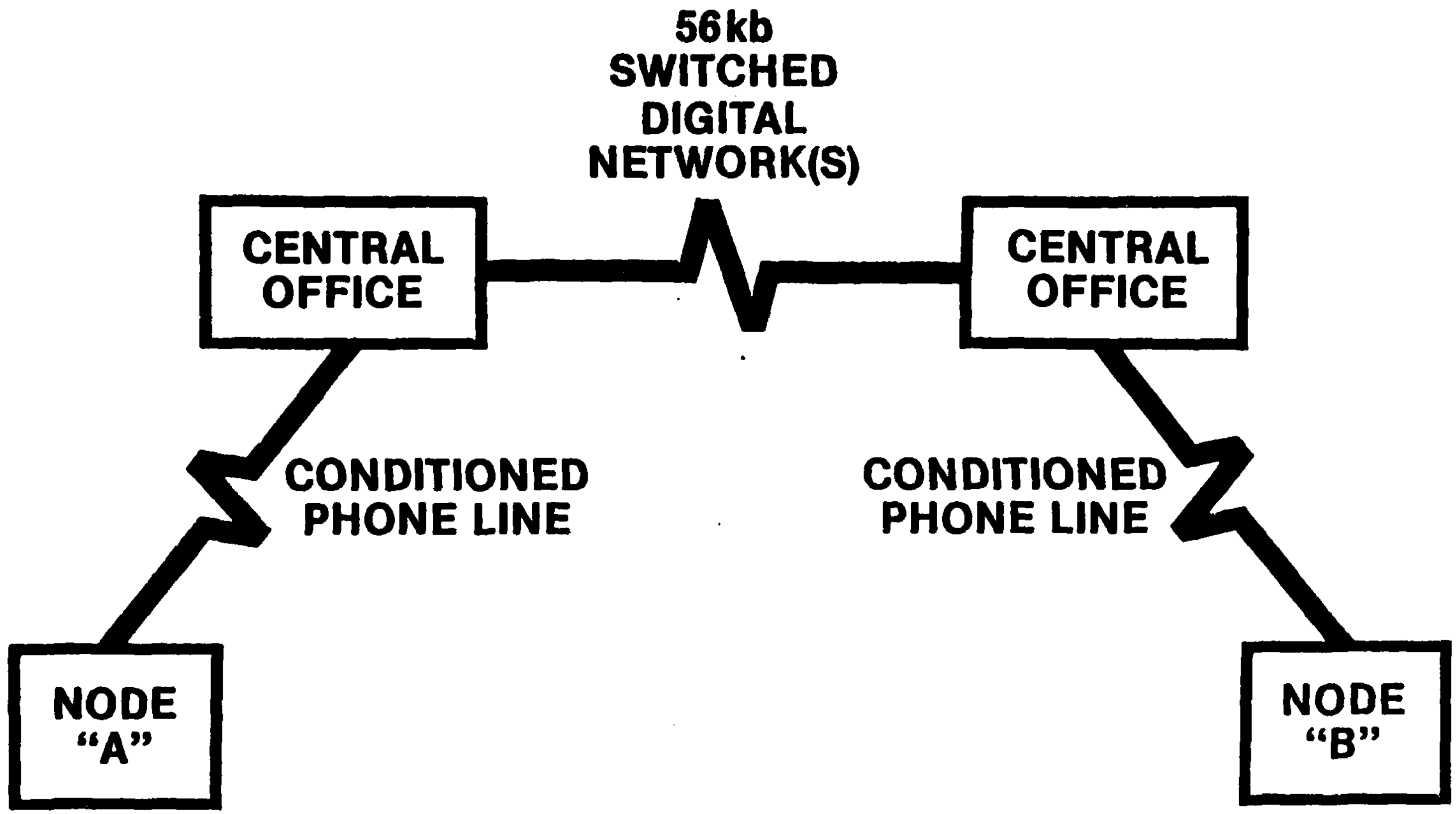
Objective: Standard Materials - Graphics and videotaped materials were transmitted through the system and the results documented at each site.

During the period of this test, August 24 through August 31, 1984, approximately fifty (50) persons, from up to ten (10) military and government agencies, were given an opportunity to evaluate this video teleconferencing system. Due to time constraints and the high status of the attendees, many of these individuals were not asked to or were not able to fill out questionnaires. However, in informal discussions with these persons, their evaluation of the system's acceptability as a management tool did not vary significantly from those who completed questionnaires.

5.1 Questionnaire Results

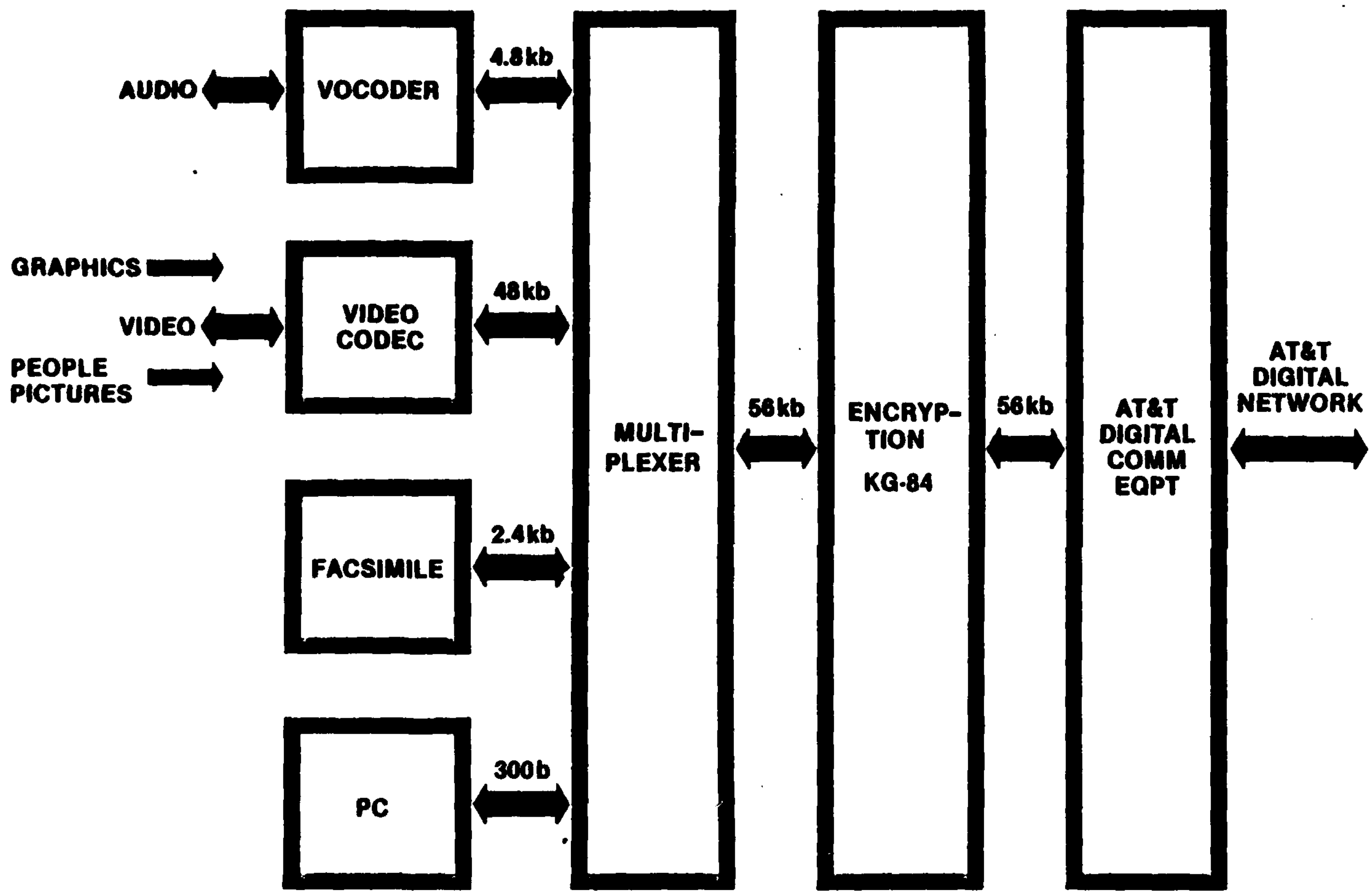
The following is a summary of the responses given in sixteen (16) participant questionniares.

How would you rate:	Highly acceptable	Acceptable	Unacceptable
1) The Audio quality?	5	7	0
2) The Video quality?	9	7	0



NETWORK CONFIGURATION

Figure 4-3



FUNCTIONAL BLOCK DIAGRAM OF TERMINAL EQUIPMENT

Figure 4-2

requested them. Copies of these two handouts are included in Figures 4-2 and 4-3. In addition to these handouts, a system overview description, developed by FEMA, was made available to test participants.

Detailed documentation on the teleconferencing system as it was fabricated and tested is included in Appendix E.

should be screened for test participants before they saw the system in operation. It was further decided that Delta Information Systems should be responsible for the development of this videotape.

During further discussions on this topic, the videotape development task was expanded to include the following:

1) An introductory tape would be developed, as specified, to explain the idea of video compression and point out some of the potential benefits of this type of system. This tape would avoid becoming too technical. A transcript of the tape that was developed appears in Appendix D.
2) A second, shorter tape would be developed, with a more technical orientation, that would deal with the system block diagram and the transmission netowrk configuration. This tape would be available for viewing by those individual participants who wanted more system details.
3) After the test/evaluation program was completed, these same videotapes would be re-edited to include some excerpts of the 56 kbps video system in operation. This would provide NCS and FEMA with "stand alone" videotapes that explain this technology and give examples of the level of video quality that the systems can deliver.

Additional Materials

The system block diagram and the transmission network diagram for 56 kbps videoconferencing were developed for the videotape and were also made available as handouts to the test participants who

feeling of the environment, were in full view. Although the codec and other support equipment was placed behind room dividers, cables were in evidence and lighting was augmented by a tripod-mounted flood. Every effort was made to minimize the visual distractions of this environment, but the facility and the necessary equipment could not be changed. Yet, despite the shortcomings inherent in the economic nature of this test, participant reactions were almost unanimously positive towards the test and the system.

4.2 Support Materials

Videotapes

When a new system or concept is being evaluated, there is always a danger that the evaluators will be unable to avoid bringing their prejudices to bear on the judgements they will be making. This possibility seemed particularly likely in the evaluation of 56 kbps videoconferencing, since the video image quality that can be transmitted at this reduced bandwidth is noticeably lower than the standard home reception video quality we are all used to.

The developers of this test program recognized that the test participants could not be expected to make responsible system evaluations unless a way could be found to give them some insight into the technology involved and to help them evaluate the potential utility of this type of system by developing a cost/benefit context for them to work from. It was decided that a videotape should be developed for this purpose, and that the tape

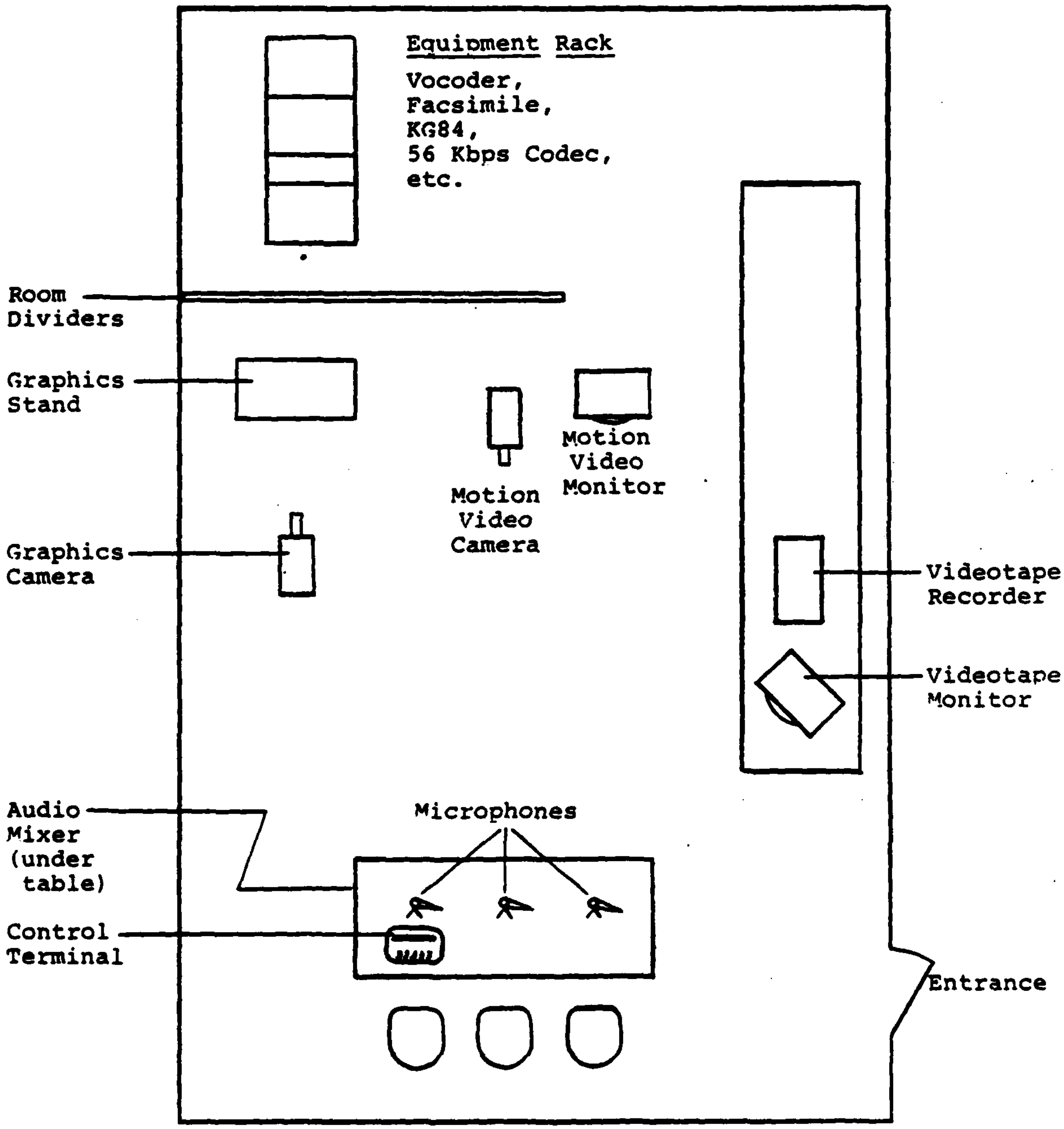


Figure 4-1 Demonstration Room Layout

4.0 Documentation

4.1 Test Environment

The site selected by FEMA for one node in the test was the FEMA Emergency and Coordination Center (EICC) at FEMA headquarters, 500 C Street, SW., Washington, DC. The EICC is a state-of-the-art "communication and information processing center linking FEMA to Emergency Operation Centers at the regional, state, and local levels across the country."

The physical set-up in the demonstration room was satisfactory, but was not that found in a fully operational teleconference room. The usual design of videoconferencing facilities includes the removal of all but the most necessary technical equipment from the conference room. This is done primarily to put users at ease, to eliminate the sense of being in a TV studio. A secondary reason for putting equipment in an adjacent room is to reduce the noise level in the conference room, since some of the equipment employs fans used to cool the components. However, this separation of equipment and conference room requires at least some minimal construction.

Because this test was to be of short duration and nominal cost, it was decided that no such construction could be justified. This meant that most of the equipment associated with this test was located in the same room with the participants. Figure 4-1 is a layout of the demonstration room which illustrates how the test equipment was arranged. Cameras and monitors, frequently recessed in videoconferencing room walls to minimize the "high tech"

for those not on camera. Allow this segment to continue and the agenda to be set at the participants' discretion.

4) Discussion and Evaluation;

When participants have satisfied their need to experience the system, begin to ask for their general evaluations and thoughts on where such systems might support their activities. Answer their further technical questions and hand out block diagrams and system description documents. Hand out Evaluation forms and ask them to complete them before they leave.

This presentation format was followed with all groups that had time enough to spend on this evaluation.

Aside from their system evaluations, the attendees were universally interested in the test and expressed their enthusiam for the project.

Graphic Materials

Although the configuration of this system did not provide an opportunity to send slides or transparencies, a number of representative graphic documents were selected for transmission and evaluated for legibility at both sites (See Figures 5-1 through 5-4). Duplicate graphics were sent in each direction for this test. The results of this evaluation are outlined in the Test and Evaluation section of this report.

Presentation Structure

A FEMA representative delivered each presentation to visiting groups at the FEMA building site. When time permitted, the following agenda was employed:

1) Introductions, Overview, and Room Orientation;
   Introduce participants to others in the LOCAL room, seat VIPs in chairs aligned to video cameras. Give a brief description of the test and of the agenda they will be following. Point out the main equipment items in the room. Introduce the videotape as an important step in making their system evaluations.
2) Play Videotape;
3) Interactive Videoconferencing;
   Turning on the picture and sound to the DISTANT site, introduce the distant participants and allow the local participants to "chat" with those at the other end. Illustrate system "graphic" capabilities, answer technical questions, offer tours of equipment areas

NCS/FEMA VIDEOCONFERENCING TEST QUESTIONNAIRE

We'd like your opinion of the videoconferencing system you have just tested. How do you weigh the balance of advantages and disadvantages?

Disadvantages:	Advantages:
Reduced Picture quality for motion images	Accessibility; 56 Kb Network many locations,
Short delay in sending graphic images	Portability; System can be used FROM many locations,
	Low transmission costs,
	Signal security.

How would you rate:	*Highly acceptable*	Acceptable	Unacceptable
1) The Audio quality?	[ ]	[ ]	[ ]
2) The Video quality?	[ ]	[ ]	[ ]

3) If you had a requirement for videoconferencing, would this type of system be useful for you?

----

4) What specific agency or department applications do you think the system has (if any)?

Name ---------------

Address ---------------

Phone ---------------

Figure 3-1

3.0 Selection of Test Materials

The following materials were developed to support this test:

Introductory Videotape

As described above, in order to assure that users would be able to make informed judgements on the acceptability of this system as a management tool, it was decided that a videotaped introduction to this technology would be developed by DIS and screened for each user group before they saw the system in operation. This tape proved to be effective in setting the context for this test, allowing participants to judge the system's quality from a more fully informed perspective.

Questionnaire

A brief questionnaire was developed and filled out by 16 user participants in the test. A copy of this questionnaire is included as Figure 3-1, and the results of this questionnaire survey are outlined in Section 5.0, the Test and Evaluation section of this report.

Codec Comparison Videotape

A Codec comparison videotape, currently under development for NCS, was transmitted through this system, and the coded result was recorded at the receive site. The coded outcome tape will provide a useful base-line document for the future evaluation of this system as compared to other codecs and other transmission speeds.

TABLE 2-1

TERMINAL EQUIPMENT SELECTED FOR THE

TELECONFERENCING TEST AT 56 KBPS

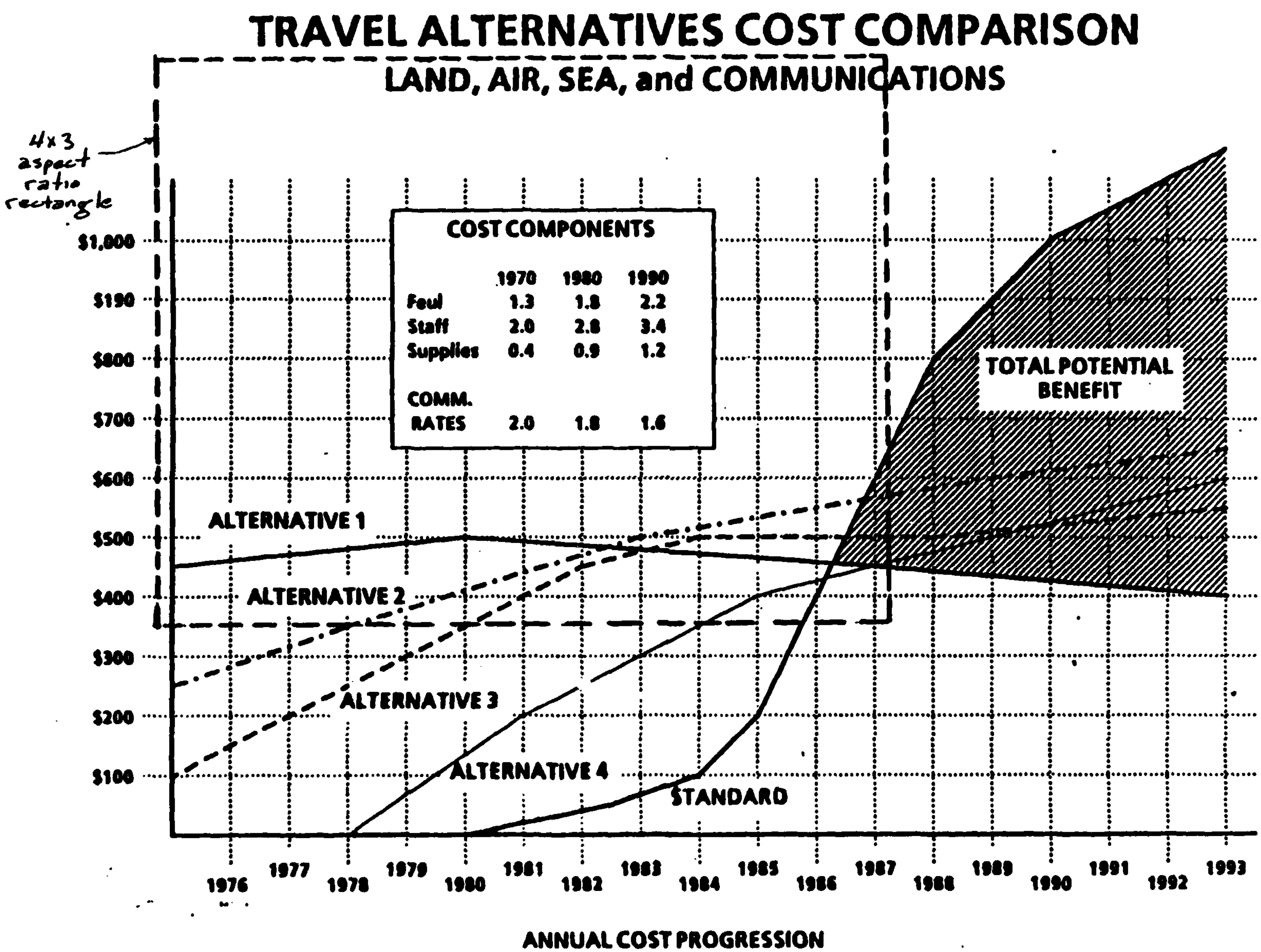
	TITLE	MANUFACTURER	MODEL NO.	QTY/NODE
1.	Multichannel Crypto Controller	Data Products	MC^3	1
2.	Vocoder	Ilex	VDC 4824T	1
3.	Video codec	Widcom	VTC-56	1
4.	Facsimile	Rapicom	3300	1
5.	Data terminal	Texas Instruments		1
6.	Crypto unit		KG-84	1
7.	56 Kbps Modem	AT&T	DSU/CSU	1
8.	TV Monitor			2
9.	TV Camera			2
10.	Audio Mixer	Shure	M268	2
11.	Audio Speaker			1
12.	Audio Microphones	Shure		2
13.	Video switch			1

graphics are displayed on a distant screen with the same resolution as any other motionless image. This image resolving capability for graphic materials is an important measurement of codec performance.

In order to develop a quantifiable measurement of system effectiveness in this respect, the sample graphic documents included in Figures 5-1 through 5-4 were transmitted between sites and the observer at the receive site requested camera adjustments (zoom and focus) until the image at the receive site was "just legible". The (4 x 3 aspect ratio) rectangle visible on the receive site screen at that "just legible" camera distance was then noted directly on the duplicate document at the receive site. This process was then repeated in the opposite transmission direction.

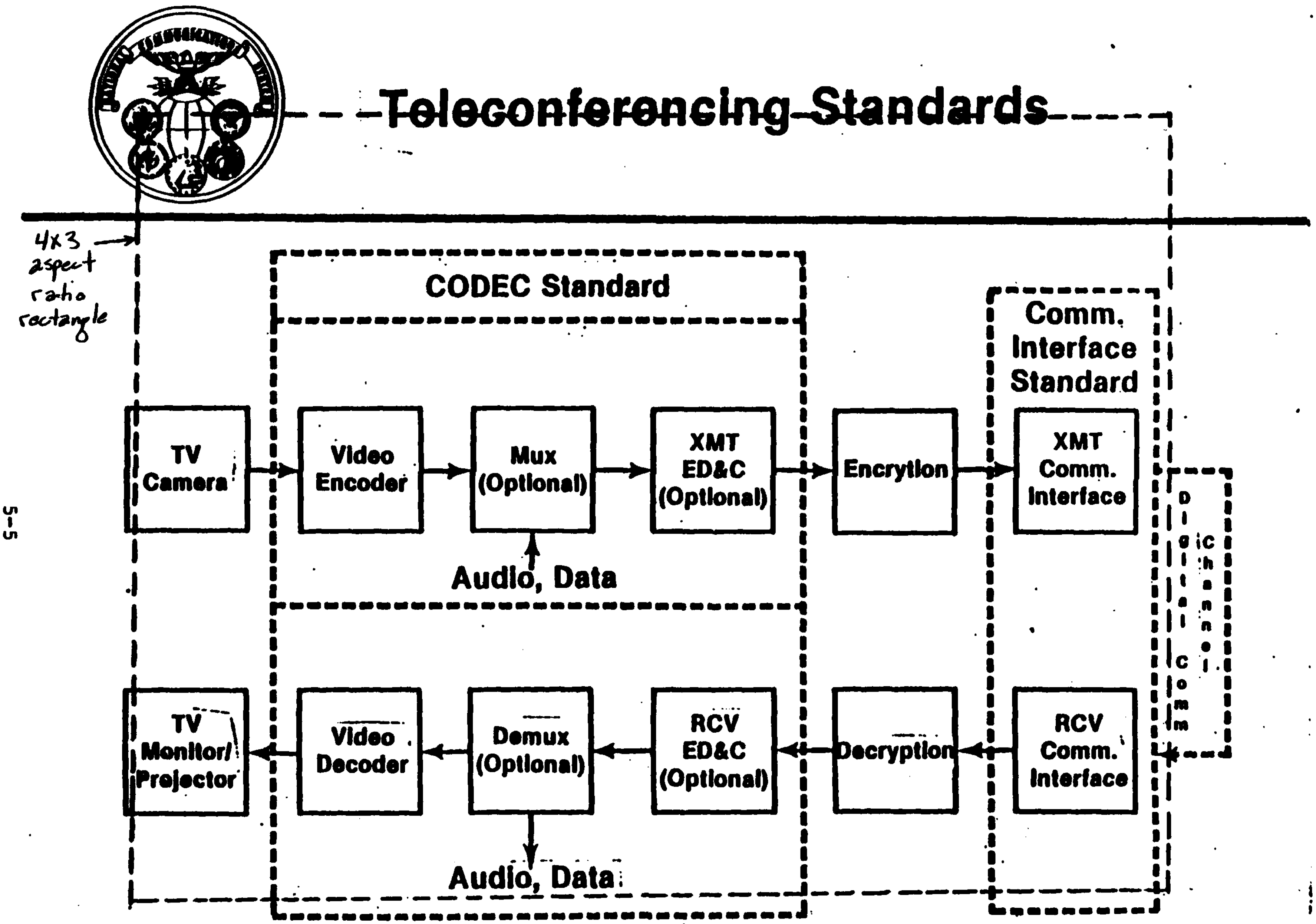
The average of these two "graphic image resolution" limits is represented by the rectangular area enclosed in the dotted lines on the xeroxed copies of test documents 5-1, 5-2, and 5-3.

Test document 5-4, illustrating various type fonts and sizes, was transmitted in both directions with the cameras adjusted so the graphic filled the entire video screen. Both sites reported the same degree of legibility on this document. The lines of type that <u>could not</u> be read at the receive sites are marked with small dots in the margins of the attached document copy.



VG 3 JUNE 1984

Figure 5-1
Test Document A



Test Document B

Figure 5-2

4x3 aspect ratio rectangle

NATIONAL COMMUNICATIONS SYSTEM

ESTABLISHMENT

The National Communications System (NCS) was established on August 21, 1963, by Presidential Memorandum to the Heads of all Departments and Agencies, entitled "Establishment of the National Communications System." The NCS is a confederation in which selected Federal departments and agencies participate with their telecommunications assets to provide essential communication services for the Federal Government under all conditions ranging from normal day-to-day situations to national emergencies and international crises, including nuclear attack. The principal assets of the NCS include telecommunications networks of the Departments of State, Defense, Interior, Commerce, Energy, and Transportation, the Federal Aviation Administration, Federal Emergency Management Agency, and the U.S. International Communication Agency, National Aeronautics and Space Administration, General Services Administration and Central Intelligence Agency.

CONCEPT

These assets comprise the bulk of the long-distance telecommunications resources of the Federal Government. Telecommunications facilities are planned, funded, and operated by the parent agencies to satisfy their respective mission requirements. However, through joint planning, standardization, and other coordinated management activities of the NCS, they are available to satisfy national requirements transcending those of the individual agencies. The objective is to ensure that essential Federal telecommunications resources are improved progressively and can be interoperated so that the aggregate functions as a coherent system.

MISSION AND FUNCTIONS

The mission of the NCS is to assure that the most critical telecommunications needs of the Federal Government can be met in any anticipated emergency situation while, at the same time, achieving the most effective yet economical fulfillment of the day-to-day telecommunications requirements.

The Manager, NCS, performs the following functions in support of the NCS mission:

a. Develops plans and procedures for the management of Federally owned and leased telecommunications assets during Presidentially declared disasters and emergencies under the Disaster Relief Act of 1974 (Public Law 93-288 and Annex C-XI, Federal Emergency Plan D).

Test Document C

Figure 5-3

Figure 5-4

Test Document D

• = not legible through system

6.0 Conclusions and Recommendations

6.1 Conclusions

While the quantifiable codec performance evaluations are valuable as a means of comparing one system with another, in the final analysis the subjective reactions of the test participants are the only reliable measure of the potential utility of this particular system.

As indicated by the questionnaire responses, there were no individual negative responses to this system. All participants found the system to be either acceptable or highly acceptable. The list of potential applications for the system also indicates a substantial perceived need for this type of communications capability in the agencies represented.

It is concluded that this type of system would find substantial utility among these user populations.

6.2 Recommendations

It is recommended that a design be developed for a generic teleconferencing node. Guidelines for the design effort are listed below:

- The transmitted bit rate will be at the nominal rate of 56 or 64 kbps.
- Full motion video will be provided using the Widcom codec or equivalent.

- Audio will be transmitted at the nominal rate of 4800 bps, providing quality equivalent to the Linear Predictive Coder.
- A multiplexer will provide optional capability for the transmission of high resolution graphics at 2400 bps and interface with a system controller at 300 bps.
- Provision will be made for encryption.
- It is important to minimize cost of the terminal equipment.

Five tasks would be performed in the design effort, as outlined below:

Task 1 - Finalization of Functional Requirement

Task 2 - System Design

Task 3 - Cost Analysis

Task 4 - Application Study

Task 5 - Final Report

APPENDIX A

MULTICHANNEL CRYPTO CONTROLLER MANUAL

MC^3 MULTI CHANNEL CRYPTO CONTROLLER

TIME DIVISION MULTIPLEXER

CRYPTO CONTROLLER

GENERAL

The MC^3 is a secure, up to 16 channels, Time Division Multiplexer which incorporates the operational requirements of a Crypto Control Unit for the KG-30 series and KG-84 Crypto Units along with automatic and manual diagnostic functions in a single space savings, cost-effective unit.

The MC^3 is listed on both the Government's Preferred Products List (PPL) and GSA Schedule.

For interfacing a KG-13 Crypto Unit, the KG-13 must be operated with a Crypto Ancillary Unit AN/UYK-22 ( ) (V) -- DNE series CAU-1000, CAU-2000 or CAU-3000. In this mode the Crypto control functions of the MC^3 are modified; the MC^3 uses the Sync Initiate control of the CAU to reframe upon loss of frame, and the Data Inhibit control to learn that crypto resync is taking place.

The MC^3 may also be conditioned for operation without any crypto unit to directly interface a Protected Wire Distribution system where encryption is not required. This operation without crypto also allows cascading MC^3 units to permit multiplexing more than 16 ports into one aggregate data stream. The trunk output of one MC^3 is connected directly (cascaded) into a synchronous port appearance of a second MC^3, which interfaces the communications facility, with or without a crypto unit. This second MC^3 can support as many cascaded MC^3's as its aggregate data rate will allow.

MC^3

Accredited by the TEMPEST Qualification Special Committee as meeting the requirements of NACSEM 5100, the Time Division Multiplexer:

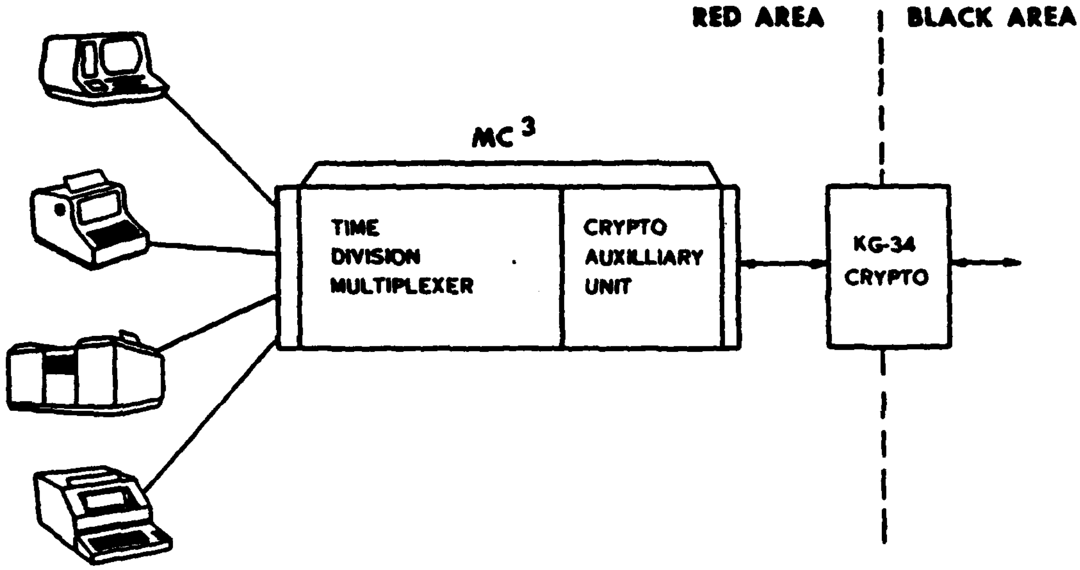
- accepts synchronous, asynchronous, and isochronous data with aggregate trunk speeds up to 64K bps
- provides for simultaneous configuration of MC^3 units at both ends of the circuit via its downline loading capabilities, and
- allows for configuration via a control terminal employing simple man/machine interchanges

The Crypto Control logic provides for:

- out of sync detection
- automatic resynchronization, and
- commanding alarm checks

The built-in Diagnostics:

- monitor system performance and provide constant status outputs, and
- provide for port and trunk loopbacks at either end of the circuit.



TDM SPECIFICATIONS

The MC^3 is a full-duplex, point-to-point, character-oriented TDM which provides for complete transparency of all data from port to port.

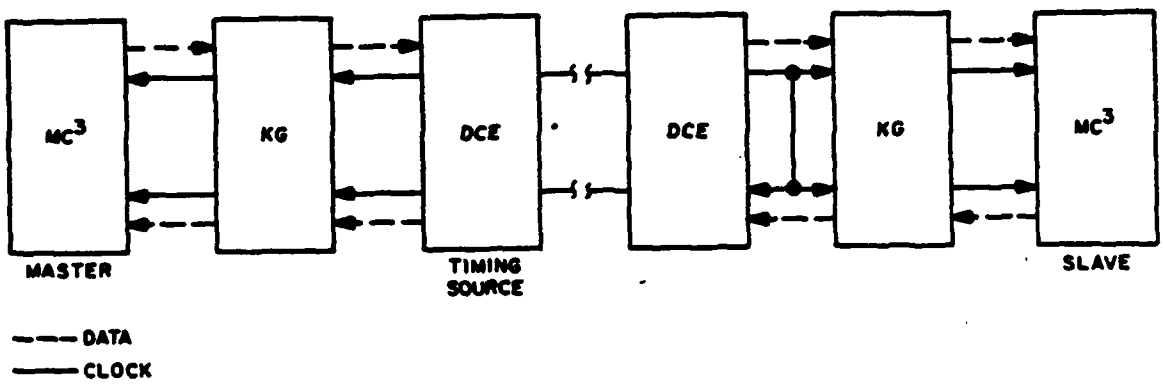
- A flexible mixture of synchronous, asynchronous, and isochronous user ports are efficiently supported.
- All asynchronous traffic uses start-stop format, with 5 or 8 data bits per character, one of which may be parity.
- The MC^3 can tolerate a continuous asynchronous port speed variation of .5%.
- All of the clocks to the ports are locked to a common timing source.

STANDARD DATA RATES

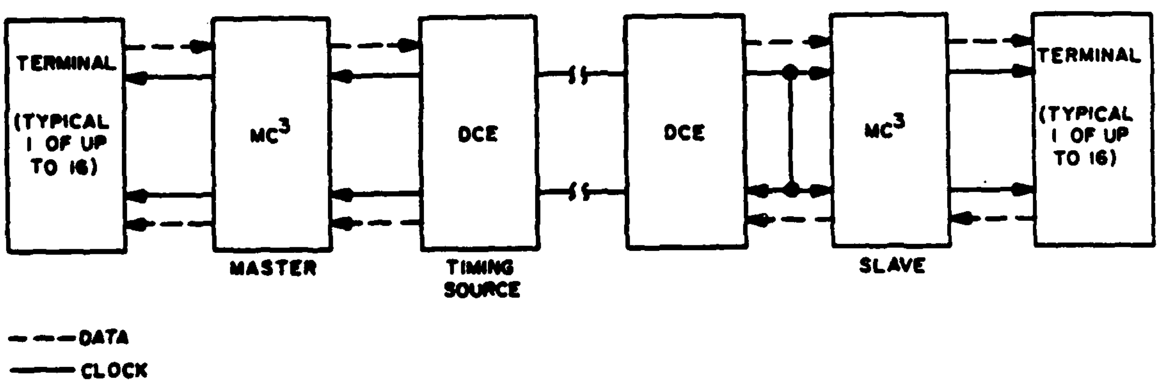
PORT RATES			TRUNK RATES
ASYNCH (bps)	SYNCH (bps)	ISOCH	SYNCH/AGGREGATE (Kbps)
75	150	0	1.2
110	300		2.4
150	600		4.8
300	1.2 Kbps		8
1200	2.4 Kbps		9.6
2400	4.8 Kbps	4000	16
4800	8. Kbps		19.2
	9.6 Kbps		32
	16. Kbps		48
	19.2 Kbps		50
	32. Kbps		56
	48. Kbps		64
	50. Kbps		
	56. Kbps		
	64. Kbps		

TIMING

- The MC^3 trunk employs a Master-Slave timing concept to provide synchronization between the MC^3 units at each end of the communications link. The timing source is on the BLACK side at the Master site. A Transmit clock is passed through the KG to the Master MC^3, which releases data at the required rate to the trunk.
- At the Slave site, the Receive clock is used on the BLACK side to furnish both Transmit and Receive clocks through the KG to the MC^3 and also serve as the Transmit clock to the trunk DCE.



MC^3 Master/Slave Timing Diagram, Cryptographic Application



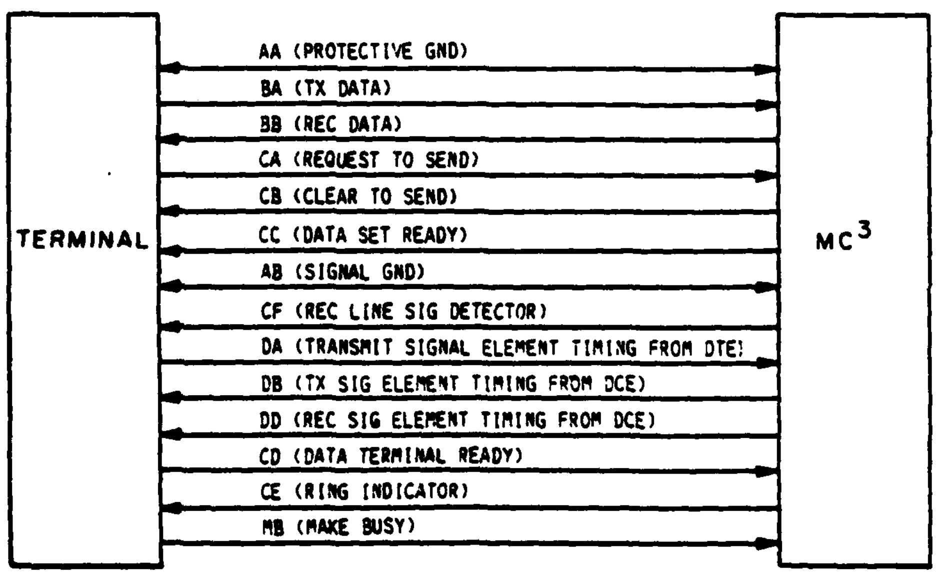
MC^3 Master/Slave Timing Diagram, Direct DCE Hookup

INTERFACES

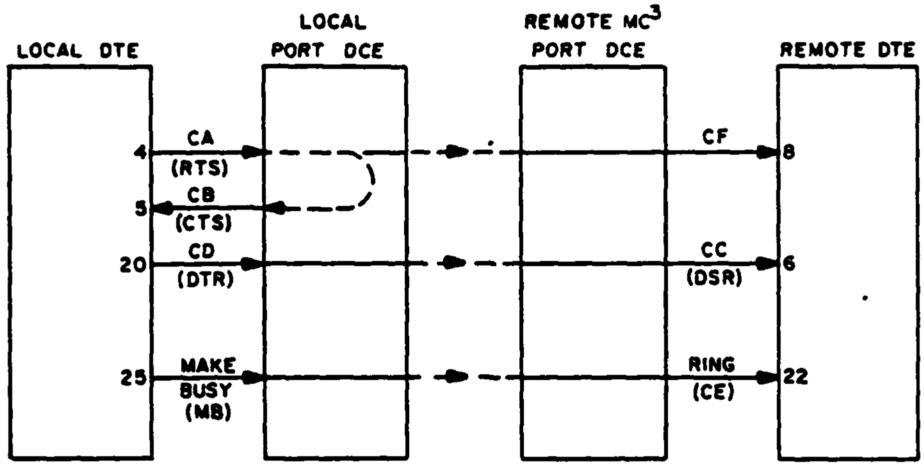
SYNCHRONOUS PORTS - Each synchronous port card is capable of full-duplex synchronous transmission. This card can accept inputs and provide outputs for two independent synchronous terminals operating at rates from 150 bps to 64,000 bps. The interface is compatible with both EIA RS-232C and MIL-STD-188.

ASYNCHRONOUS PORTS - Each asynchronous port card contains two ports which will handle asynchronous data (i.e., data framed with start and stop bits) at rates from 75 bps to 4800 bps. The asynchronous port card contains jumpers that permit the user to select either low-level polar (RS-232C/MIL-STD-188 compatible) or current loop as an interface choice.

ISOCHRONOUS PORTS - Each isochronous port card contains two ports which will handle isochronous data at rates up to and including 4000 bps. Each port is designed to accept a digital data stream which can be synchronous without clock, asynchronous at any rate up to the assigned port rate or a series of variable length bits. (If variable length bits are employed, the shortest bit element cannot exceed the maximum bit rate assigned to the port.) These isochronous port cards are similar to the synchronous port cards except the isochronous ports do not employ clock signals. The isochronous ports employ transitional encoding/decoding which require a clock frequency which is at least four times the port rate. The available clock rates (bps) are: 300, 600, 1200, 2400, 4800, 8000, and 16,000. These rates correspond respectively to isochronous port rates which are less than or equal to the following: 75, 150, 300, 600, 1200, 2000, and 4000 bps. (For example, a terminal of 87 bps requires a clock rate of at least 600.) The interface is compatible with both EIA RS-232C and MIL-STD-188.



MC³ CONTROL SIGNALS



NOTE: ONLY THE LOCAL-TO-REMOTE DIRECTIONAL SIGNALS ARE SHOWN: THE SAME SIGNALS ARE SIMULTANEOUSLY PASSED IN THE OTHER DIRECTION

CONTROL SIGNALS

Three EIA-type control signals are passed in each direction for each port, as further explained below, by port type. To protect these signals from trunk errors, a change in control signal level at one end must be received twice at the distant end before action is taken.

The MC^3 presents a DCE interface on the port side, since it is intended to connect to terminals. If it is desired to connect a tail circuit modem to a port, a crossover cable is used to convert the port to a DTE.

The MC^3 "crosses" the following pairs of control signals: DSR and DTR, RTS and CF, Ring and Make Busy. (For example, when DTR is raised by a user, DSR is sent to the remote user.) Unless the MC^3 is performing a resynchronization, CTS follows RTS.

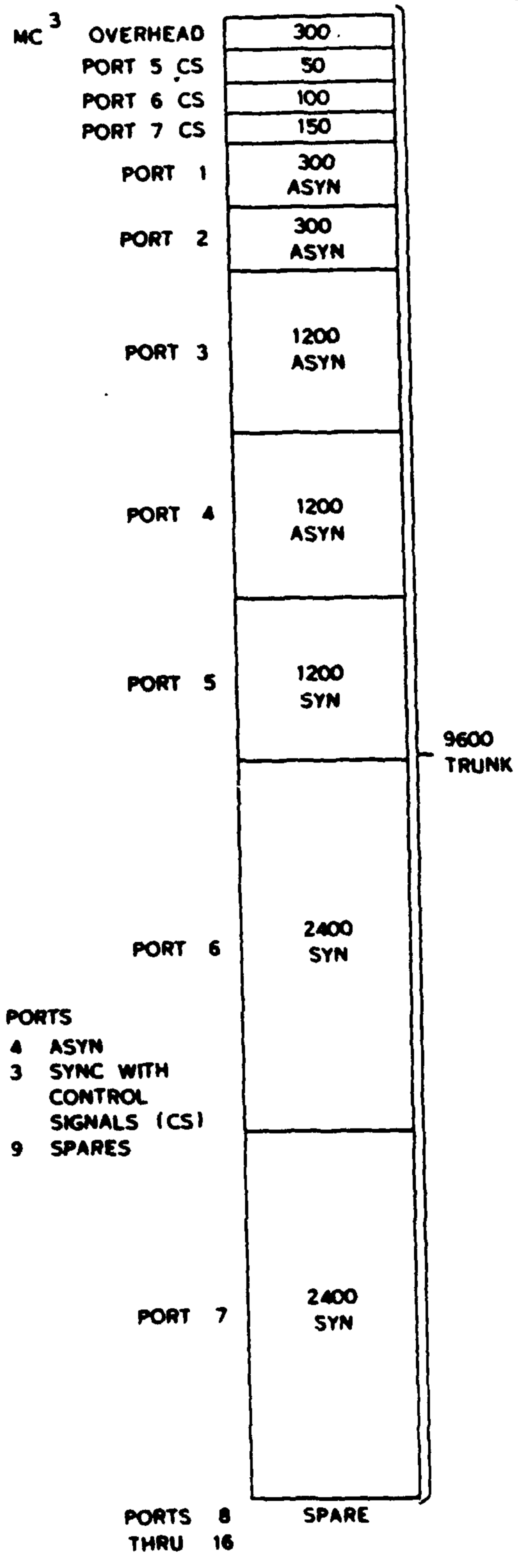
The control signals for asynchronous ports are automatically passed by interleaving them with data characters.

Controls are not automatically passed by synchronous and isochronous ports. In order to pass controls, an additional time slot(s) is configured in each frame; by specifying multiple control time slots, the response time to changes in control signals can be reduced at the expense of increased "overhead".

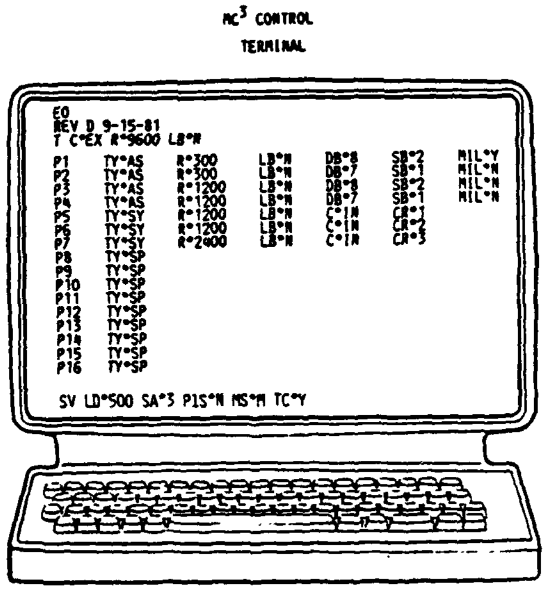
OVERHEAD CONTROL CHANNEL

The overhead control channel carries loopback commands, remote alarms, and other operator commands. It is used to detect loss of TDM framing, loss of crypto synchronization, and loss of bit count integrity. Error protection on this channel is provided to safeguard these vital functions, preventing unwanted loopbacks, etc., due to trunk errors. This provides a significant advantage over standard TDM's.

This overhead is obtained from one of two sources. Port No. 1 if it is a sync port can be caused to operate slightly below nominal speed or excess trunk capability can be used for this purpose. Several possibilities are discussed below.



CONTROL TERMINAL DISPLAY



Symbol	Definition
EO	EXAMINE OFF-LINE MEMORY
T	TRUNK CONFIGURATION
C	CLOCK SOURCE
EX	EXTERNAL CLOCK
R	DATA RATE
LB	LOOPBACK STATUS
N	NORMAL
P1	PORT ONE CONFIGURATION
TY	PORT TYPE
AS	ASYNCHRONOUS
DB	NUMBER OF DATA BITS
SB	NUMBER OF STOP BITS
MIL	IS TERMINAL MIL-188C?
SY	SYNCHRONOUS
IN	INTERNAL CLOCK
CR	CONTROL SIGNAL RATE
SV	SYSTEM VARIABLES
LD	LINE DELAY (MSEC)
SA	CRYPTO SYNCHRONIZATION ATTEMPTS
P1S	IS PORT ONE SLOWED?
MS	MASTER OR SLAVE?
TC	TRUNK CONTROLS
SP	SPARE

LIST OF FIGURES

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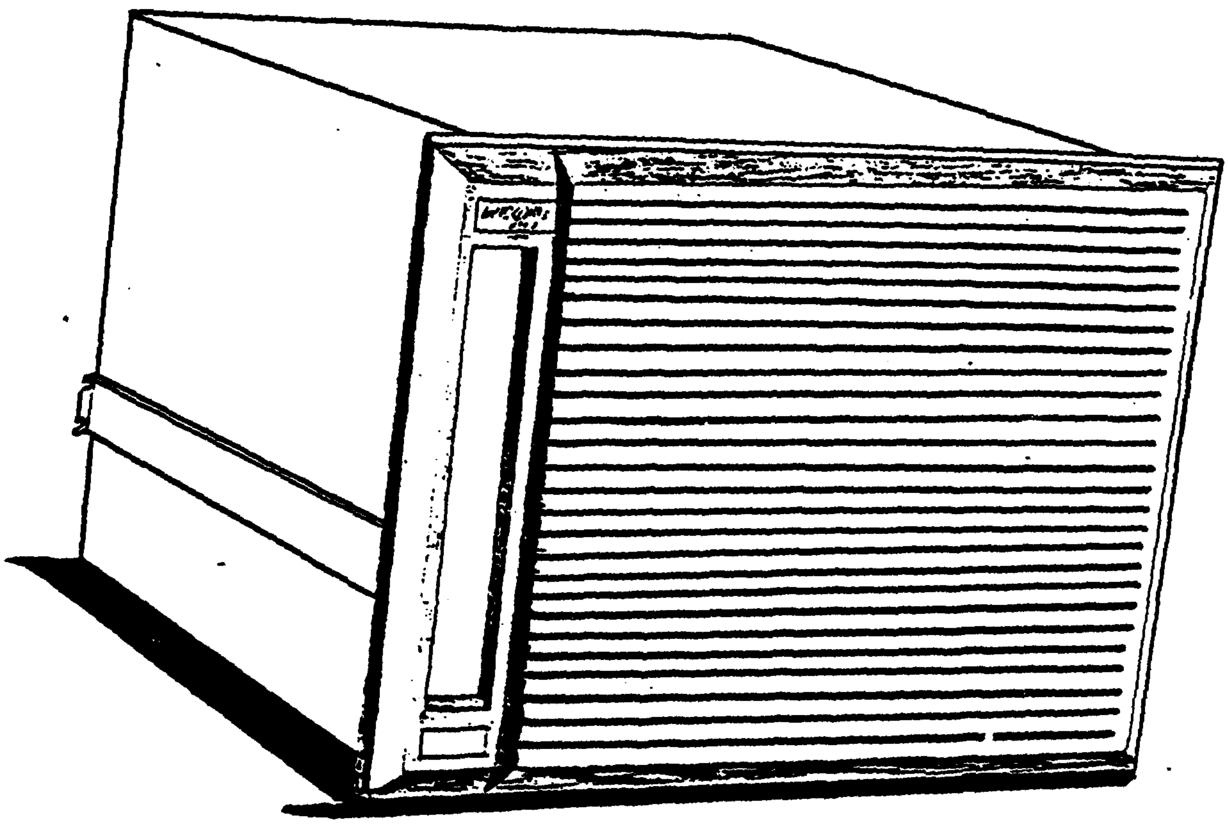
PRELIMINARY



WIDCOM
VTC-56

WIDCOM INCORPORATED
Campbell, California

APPENDIX B

WIDCOM VTC-56 CODEC INSTRUCTION MANUAL

MC^3

MULTICHANNEL CRYPTO CONTROLLER

ORDERING INFORMATION

(Listed on PPL and GSA Schedule)

The MC^3 consists of a basic unit (which includes chassis, power supply, and common logic cards) and dual port cards which must be ordered separately. Each dual port card interfaces two ports of the same type, i.e., synchronous, asynchronous, or isochronous. A maximum of eight port cards (16 channels) may be inserted into one basic unit. When ordering, specify type and quantity of each port card required along with basic unit.

Description	Part Number
MC^3 Basic Unit	93251010-001
MC^3 Sync Card	83251000-000
MC^3 Async Card	83251010-000
MC^3 Isoc Card	83251020-000
Operator's Manual	24000625-000

Table 6-1. Multichannel Crypto Controller - P/N 93251010-001

Description	Part Number
Power Supply	48070117-000
Fan Assembly	48090042-000
Trunk Card	83251050-000
Harness Board	83251070-000
Front Panel Card	83251080-000
Control Terminal Card	83251090-000
Shared Logic Card	83252020-000
Receiver Processor Card	25600008-001
Transmitter Processor Card	25600008-002
Filter Assembly	93251025-000

CONTROLS

1 Power toggle switch applies primary power to the unit's power supply unit.

2 A 3-position Trunk Loopback switch selects among Normal (unlooped), Local Trunk Loopback, and Remote Trunk Loopback requires subsequent activation of Trunk Actuate switch to effect a change.

3 A momentary toggle Trunk Actuate switch forces the unit to attempt to acquire frame synchronization on the trunk.

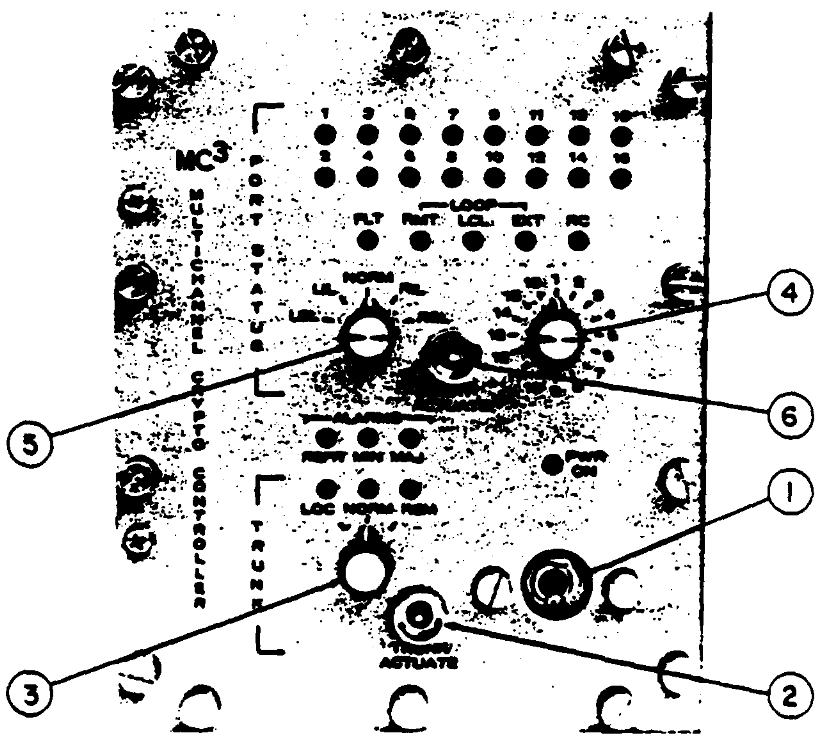
4 A 16-position rotary switch (Port Selector switch) is used to designate a port to be operated upon by the Port Loopback switch described below. The detail status of a particular port is selected by this rotary switch.

5 A 5-position Port Loopback switch selects the loopback status of the individual port designated by the Port Selector switch described above. It selects among Local External Loop, Local Internal Loop, Normal (unlooped), Remote Internal Loop, and Remote External Loop.

6 Port Actuate switch is used to engage or cancel port loopbacks as selected in 4 and 5.

OTHER CONTROLS

- The Control Terminal port is strap-selectable among several standard rate/format combinations (110, 300, 1200, 2400 bps).
- The Master/Slave selection is accomplished via the Control Terminal.



CONTROLS AND INDICATORS

The main MC^3 *front panel holds* controls and indicators related to the ports, the trunk, and the entire unit.

INDICATORS

A A green power (PWR ON) LED - ON whenever primary power is applied to the MC^3 power supply input.

B Major (MAJ) Alarm LED - ON when an outage or malfunction is sensed which prevents communication on all ports of the MC^3. This occurs after the user-selected number of resync attempts is exceeded.

C *Minor (MIN) Alarm LED* - ON when a fault is sensed which prevents normal operation of one or more of the individual ports.

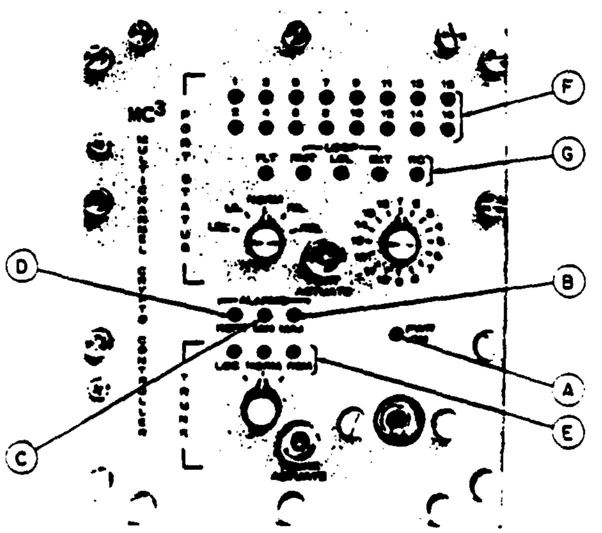
D Reframing (REFR) LED - ON when the MC^3 starts its reframing procedure, and is turned OFF when frame synchronization is successfully acquired.

E Three LED's indicate the Trunk Loopback status (Local, Normal, or Remote). The true status is displayed, which may differ from the position of the Trunk Loopback switch <u>if</u> the status has been changed via the Control Terminal or the remote unit.

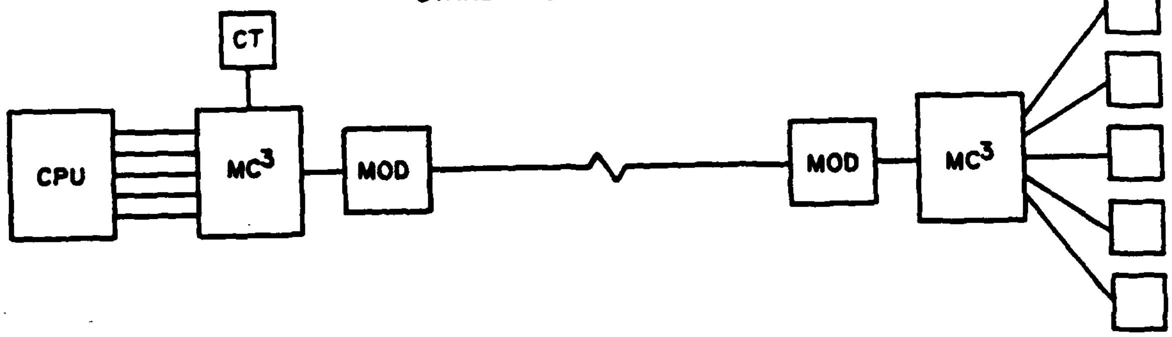
F Port Status (1 thru 16). The *front panel* holds 16 RED alarm indicators, one associated with each port. When a fault or loopback condition exists on a port, the corresponding alarm is ON.

G *Five LED's display* the status of the port selected by the Port Selector switch.

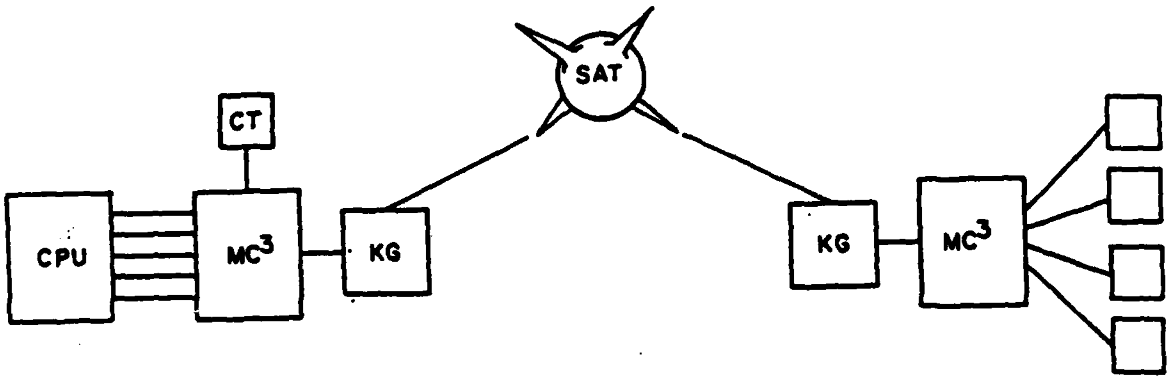
- Fault (FLT). A condition (other than loopback) has been sensed which prevents normal operation of this port.
- Remote Port Loopback (RMT)
- Local Port Loopback (LCL)
- External (EXT). An External Port Loop is in progress.
- Remote Control (RC). The latest change to the loop status *of* the local port originated at the remote MC^3.



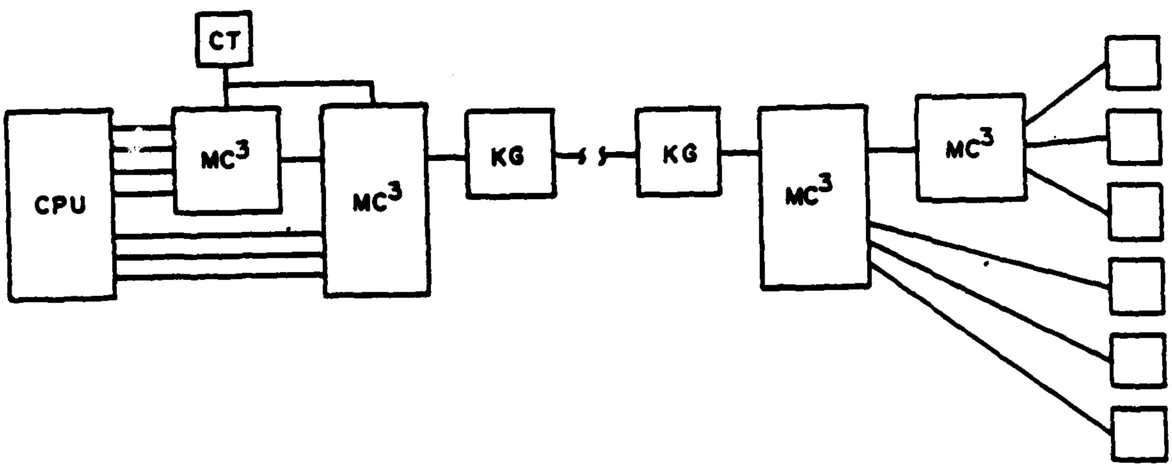
STAND ALONE BLACK TDM



SATELLITE OPERATION

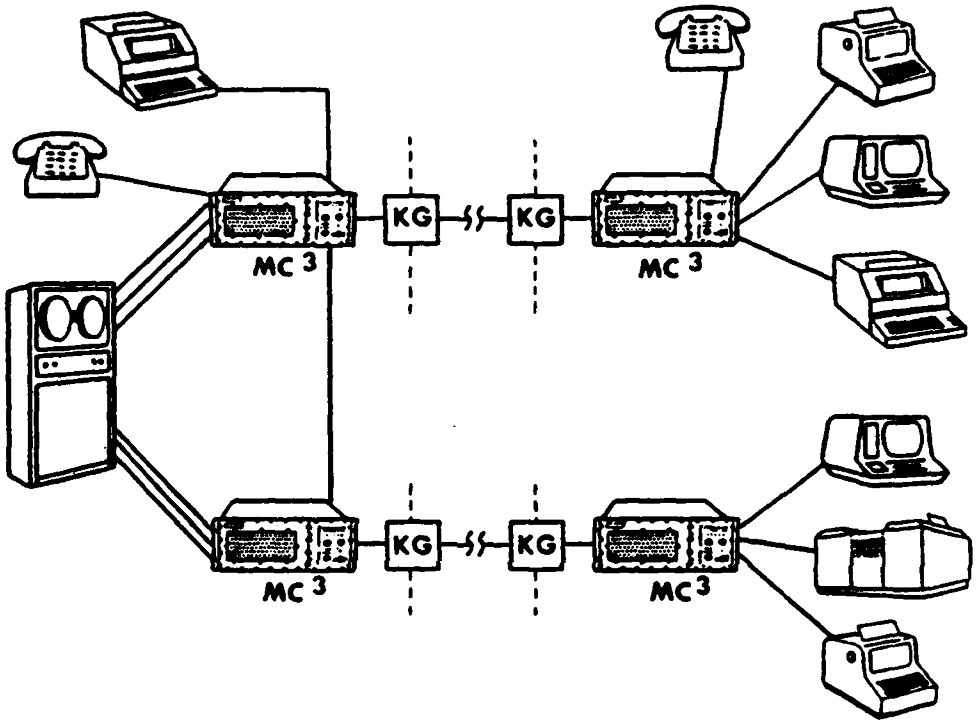


TANDEM MULTIPLEXER



APPLICATIONS

The MC^3 flexibility permits application in a variety of secure communications networks. Several examples are displayed below:

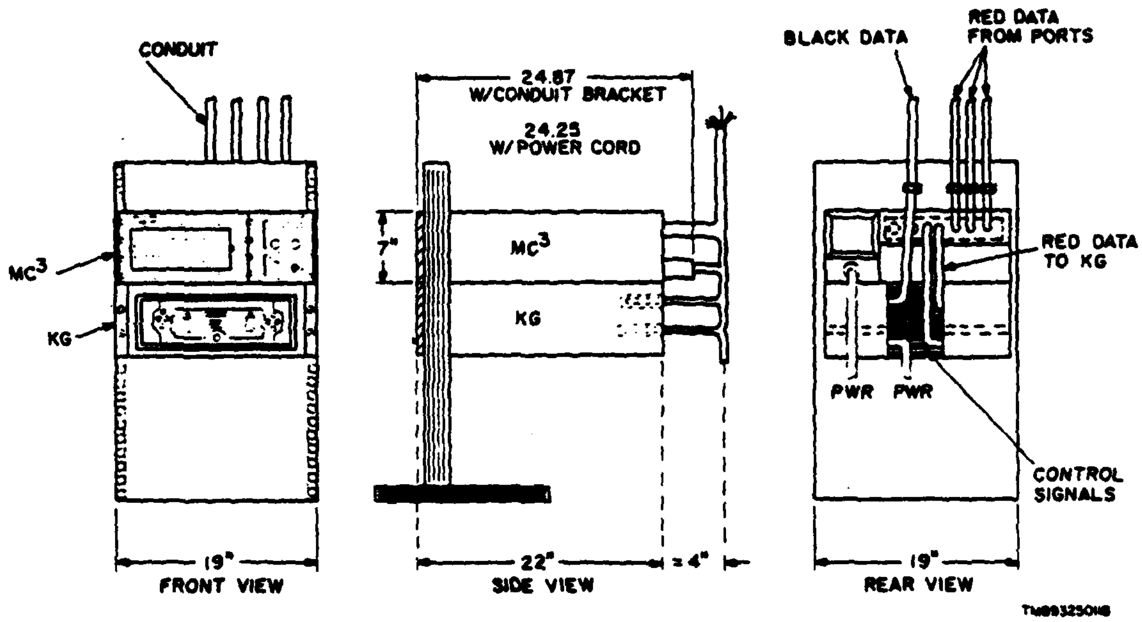


The ability to support digital voice, data, facsimile, and record traffic at a variety of rates provides for the integration of different types of information. The whole system may be operated and diagnosed from one central site by sharing one control terminal between a number of MC^3's. MC^3 bulk encryption replaces multiple cryptos, crypto auxiliary units, and data communication equipment. For example, the utilization of MC^3 with only four 2400 synchronous ports would result in substantial savings in hardware and line costs.

INSTALLATION

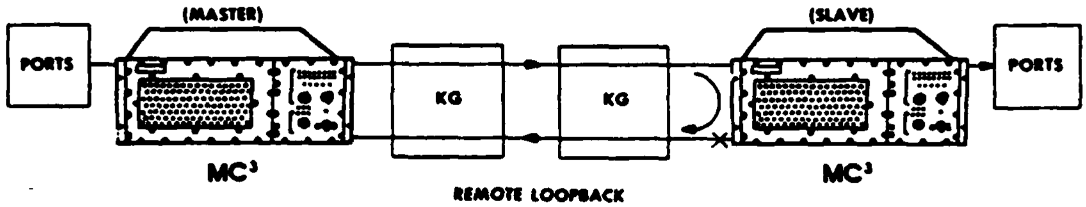
The MC^3 is designed to be installed in a standard 19-inch rack and will consume 7 inches of vertical rack space. All connections are via the back panel which will accept conduit fittings. Port connectors accept standard RS-232 25-pin connectors; the aggregate trunk to the crypto connects to a 30-lug barrier strip.

The MC^3's compact size (19" x 7" x 25") allows up to 16 circuits to be bulk encrypted using a minimum of rack space.



TRUNK LOOPBACKS

In Remote Trunk Loopback of a Slave MC^3, the data and clock from the Master MC^3 are returned as above. The data is also accepted by the Slave receiver and distributed to the ports. Thus, the Master transmitter is simultaneously connected to two receivers; if one receiver operates properly and one fails, the fault is isolated to the failed receiver, while if both receivers fail, the fault is at the transmitter.



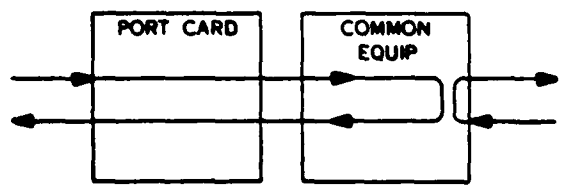
PORT LOOPBACKS

Two kinds of port loopbacks are provided: External and Internal. Both are bidirectional, may be initiated (and cancelled) either locally or remotely, and result in the user's data being echoed back to his equipment.

In External Port Loopback, the data (and clock, if any) of an individual port are returned to the user after passing through the interface receivers and drivers of the port card. This verifies the user's equipment, interconnecting cable, and the port card's interface hardware. Data from the remote unit is passed through the receiver section and then the transmitter section of the port card, returning to the remote MC^3.

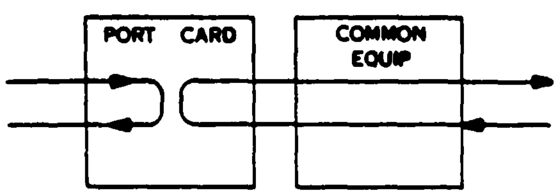
In Internal Port Loopback, the port data is passed entirely through the transmitter section of the port card, looped in the common equipment, and passed through the entire receiving section of the port card, thereby completely verifying operation of the port card. Data from the remote unit is looped in the common equipment without passing through any part of the port card.

INTERNAL (LOCAL AND REMOTE CONTROLLED)



PORT LOOPBACKS

EXTERNAL (LOCAL AND REMOTE CONTROLLED)



STATUS COLLECTION

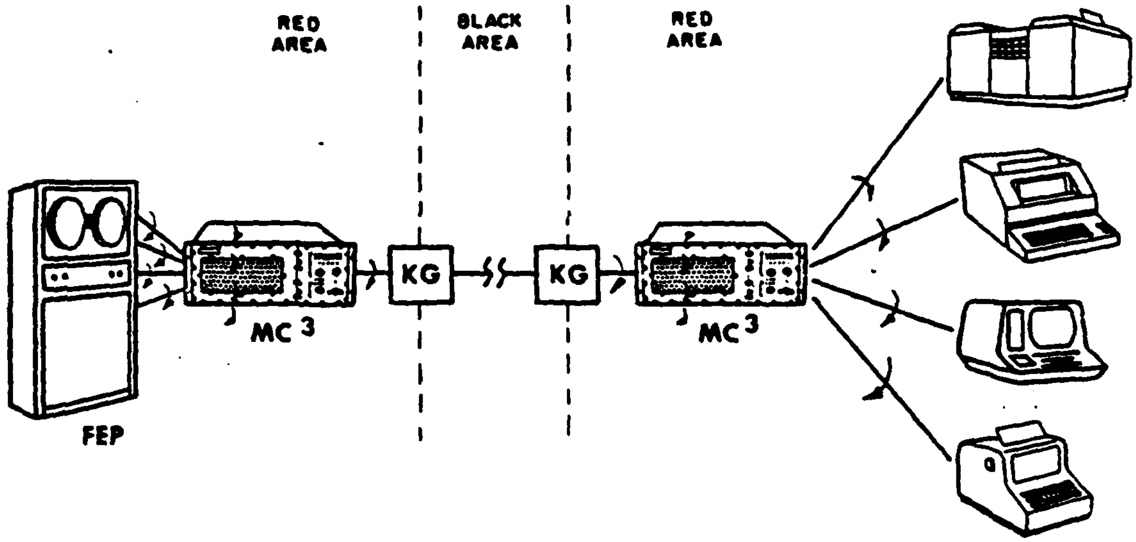
An approximate measurement of the trunk error rate is provided for each of the two directions of transmission. Other information including status of port operation is also collected. The results of these measurements are available via the Control Terminal.

DIAGNOSTICS

LOOPBACKS

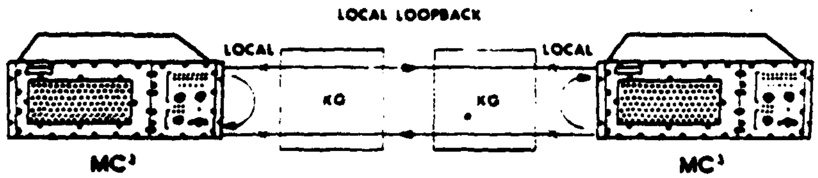
Loopback of the trunk and each individual port is provided. These loopbacks may be set and cleared at the local or remote MC^3 from the control terminal or from the front panel. Possible locations of MC^3 loopbacks are shown:

Loopback of an individual port does not disturb traffic on the other ports, although a loopback of the trunk causes MC^3 resynchronization, and loopback of all ports.



TRUNK LOOPBACKS

In Local Trunk Loopback of either a Master or Slave MC^3, an internal timing source is used. Trunk data is looped between the trunk interface circuitry and the remainder of the trunk hardware. The trunk signal from the remote unit is ignored to completely isolate the unit under test. Data entered at any local port is routed through the entire transmit section of the MC^3, then through the entire receive section and returned to the port, appearing as an echo of the input data. In this manner, the operation of the MC^3 is fully verified.



CAU SPECIFICATIONS

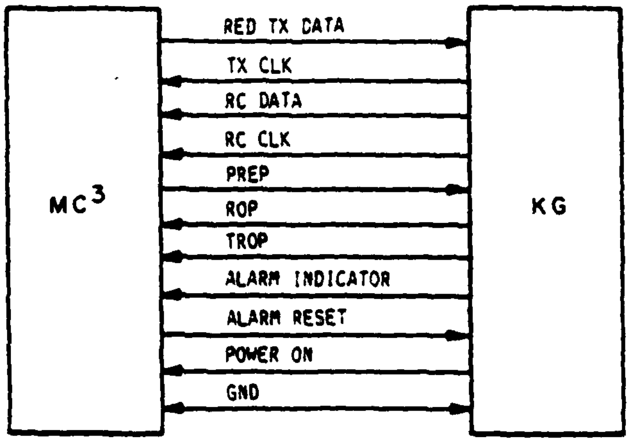
The MC^3 operates with a crypto of the KG-30 series or a KG-84. Operating completely on the RED side, it is designed to meet the requirements of NACSEM-5100 for RED equipment.

The MC^3 duplicates the functions of the 2100 or 3100 series CAU which are required by a KG-30. It performs out-of-synchronization detection and automatic resynchronization, as well as the alarm check function. It inhibits data to all TDM ports appropriately.

The MC^3 accepts CF and does not resynchronize while this input is low. This signal may be passed from the trunk modem through a separate isolator to the MC^3, or it may be strapped high at the MC^3.

The overhead channel also allows loss-of-crypto synchronization detection, without the necessity of intermessage gaps or the periodic insertion of special characters into the user's data stream. This feature is a substantial advantage of the MC^3 over separate TDM and CAU equipments.

As an option, the MC^3 can be configured as a "standalone" RED multiplexer. In this mode, it can be operated with an AN/UYK-22 CAU, which may interface with a KG-13. The MC^3 uses the Synchronization Initiate control of the CAU to reframe upon loss of frame, and the Data Inhibit control to learn that crypto resync is taking place. The interface signals between the MC^3 and a KG-30 crypto are shown:



PORT FLEXIBILITY

Up to 16 user ports are supported. One card can support two user ports of the same type (sync, async, isoc).

An efficient intermix of sync, async, and isoc ports with speeds from different families (75 x 2^N and 8000 N) is possible.

Example 1. One 8000 sync, one 1200 async, and four 110 async ports can be combined into a 9600 bps trunk.

Example 2. One 9600 sync, one 2400 sync, four 300 async, and four 150 isoc ports fit into a 16-kilobit trunk.

The above examples also show the two methods of generating the overhead bit pattern. In Example 1 above, the synchronous port must be slowed, while in Example 2, the 400 bps of aggregate bandwidth which is unused by the ports is used as overhead.

Port slowing is possible when DTE (CPU or terminal) equipment is operating with the external clock provided on that channel by the MC^3 and is capable of working in synchronization with a clock which is less than the expected rate.

CONFIGURATION CONTROL

Configuration of the TDM is done by temporarily connecting a user-supplied asynchronous ASCII terminal to a designated connector. Interactive responses are provided by the MC^3:

- to allow the user to interrogate the configuration,
- to modify it (such as by changing the port's speed),
- to set or clear loopbacks,
- to examine previously collected statistical measures of performance,
- to monitor alarm status
- to initiate the resynchronization procedure, and
- downline load a new configuration to the remote MC^3.

Once entered, configuration information is stored in non-volatile memory in each MC^3. The configuration may be established or changed from either end of the link. At the time of the next operator-initiated resynchronization, the new configuration is downline-loaded to the remote unit and becomes the active configuration in both units. No configuration changes (other than setting and clearing of individual port loopbacks) are allowed on-line. A resynchronization, involving a loss of all user's data for a brief interval, is required to activate other configuration changes. See sample control terminal display and corresponding bandwidth allocation.

1. VTC-56 SYSTEM SPECIFICATIONS

1.1 VIDEO SPECIFICATIONS

Input video source:	NTSC composite video (EIA RS-170)
Output video supplied:	2 ea. NTSC composite video (EIA RS-170)
Gen-lock Input:	NTSC composite video (EIA RS-170)

1.2 AUDIO DELAY SPECIFICATIONS

Input Characteristics:	Input impedance 10K ohm balanced, 5K ohm unbalanced. Acceptable input level for full dynamic range -9 to +12 dbm. (Note: 0dBm is defined as 3.16 volts peak to peak.)
Output Characteristics:	Output impedance 600 ohm balanced, 300 ohm unbalanced. Non adjustable Output = 12dBm when input level is at 0dB.
Dynamic Range:	78dB
Analog to Digital Transfer Function:	A law (linear approximation of logarithmic law).
Frequency Response:	300 Hz to 3000 Hz \ .125dB Down -.35dB at 3300 Hz, -.7dB at 3400 Hz and - 14dB at 4000 Hz.
Time delay:	Zero to 1.01 seconds in 32.6 milli-second steps.
Filters:	Switched capacitor filters (integrated into Intel's 2913 codec integrated circuit).
Input Level Indicators:	10 LED indicators in 3dB steps from -24dB to +3dB.
Delay Length Indicators:	32 LED indicators in 32 milli-second step from 0 to 1 second delay.
Inputs:	Line input and remote delay length switch input.

Outputs:	Line output.
Controls:	Input level adjust, bypass or delay switch, increase or decrease delay switch.

1.3 SOURCE POWER REQUIREMENTS

Input Power:	120 volts +- 10%, 60 Hz at aprox 9 amps.
	The rear panel has a 10 amp resettable circuit breaker for protection from input power faults.

1.4 ENVIRONMENTAL REQUIREMENTS

Operating Temperature:	0 to 50 degrees Celsius
Storage Temperature:	-10 to 70 degrees Celsius

1.5 MECHANICAL SPECIFICATIONS

Shipping Weight:	85 lb. (38.6 kg)
Operating Weight:	75 lb. (34 kg)
Overall Dimensions:	19" wide X 17.5" high X 24" deep Standard RETMA rack mounting slides are provided for mounting.

2. VTC-56 SYSTEM OPERATING INSTRUCTIONS

2.1 INSTALLATION

The VTC-56 Codec is designed to be mounted in a standard RETMA rack. Because of the weight of the unit it is recommended that the VTC-56 Codec be mounted in the lower half of the rack.

For proper cooling, the user should mount the VTC-56 Codec in such a manner that ensures uninterruptted air flow to both the front and rear of the unit.

2.2 POWER USAGE

The VTC-56 Codec requires up to 1200 watts at the upper limits of line voltage. Actual power consumption is dependent upon the number of installed options.

2.3 INPUT/OUTPUT CONNECTIONS

As shown in fig 2.1 all connections to the unit are at the rear panel. All connections should be made before applying power to the unit.

The video connections are standard BNC type. The recommended coax for all video signals is RG-59/U. For optimum performance all coax runs should be as short and noise free as possible.

The audio input, output and delay connections are made through standard professional audio type connectors. The recommended cable for these connections is three wire, shielded cable with at least 50% shield coverage.

The Channel Interface connection to the 56 kbaud modem is through a standard CCITT V.35 connector such as a Winchester #MRA34GJ or equivalent. It is recommended that the cable used for the inter-connection be shielded in order to reduce EMI/RFI emissions.

2.4 BOARD PLACEMENT

Refer to Figure 2.2 for proper board placement.

NOTE: If an option board is removed, an Airflow Management Board must be installed in its place to maintain proper cooling for the rest of the system.

2.5 REMOTE AUDIO DELAY OPERATION

The remote audio delay control is provided so that the user can adjust the audio channel to achieve proper lip sync with the Codec's video channel. Proper operation is acheived by increasing or decreasing the audio delay while viewing the received picture.

For detailed information refer to Section 3.11 of this manual.

2.6 EMI/RFI EMISSIONS

WARNING - This equipment generates, uses, and can radiate radio frequency energy and if not installed and used in accordance with the instructions manual, may cause interference to radio communications. It has been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC Rules, which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area is likly to cause interference, in which case the user, at his expense, will be required to take whatever measures may be required to correct the interference.

2.7 OPERATORS/SERVICE SAFETY SUMMARY

2.7.1 POWER SOURCE

This product is intended to operate from a power source that will not apply more than 125 volts rms between the supply conductors or between the supply conductor and ground. A protective ground connection by way of the grounding conductor in the power cord is essential for safe operation.

2.7.2 GROUNDING THE VTC-56 CODEC

This product is grounded through the grounding conductor of the power cord. To avoid electrical shock, plug the power cord into a properly wired receptacle before connecting any input or output cables.

2.7.3 USE CARE WHEN SERVICING WITH POWER ON

Dangerous voltages exist at several points inside the VTC-56. To avoid personal injury, do not touch exposed connections and components while power is turned on.

3. SYSTEM ARCHITECTURE

A block diagram of the system is shown in Figure 3.0-1. The encoder, shown in Figure 1-1.a, accepts a standard composite video signal meeting NTSC specs and generates a digital bit stream at 56,000 bits per second using the algorithm described in Reference 1. This bit stream carries all the information needed to recreate the original video signal with minimal distortion. The decoder shown in Fig. 1-1.b accepts the bit stream and performs the complementary decoding action, regenerating the NTSC signal.

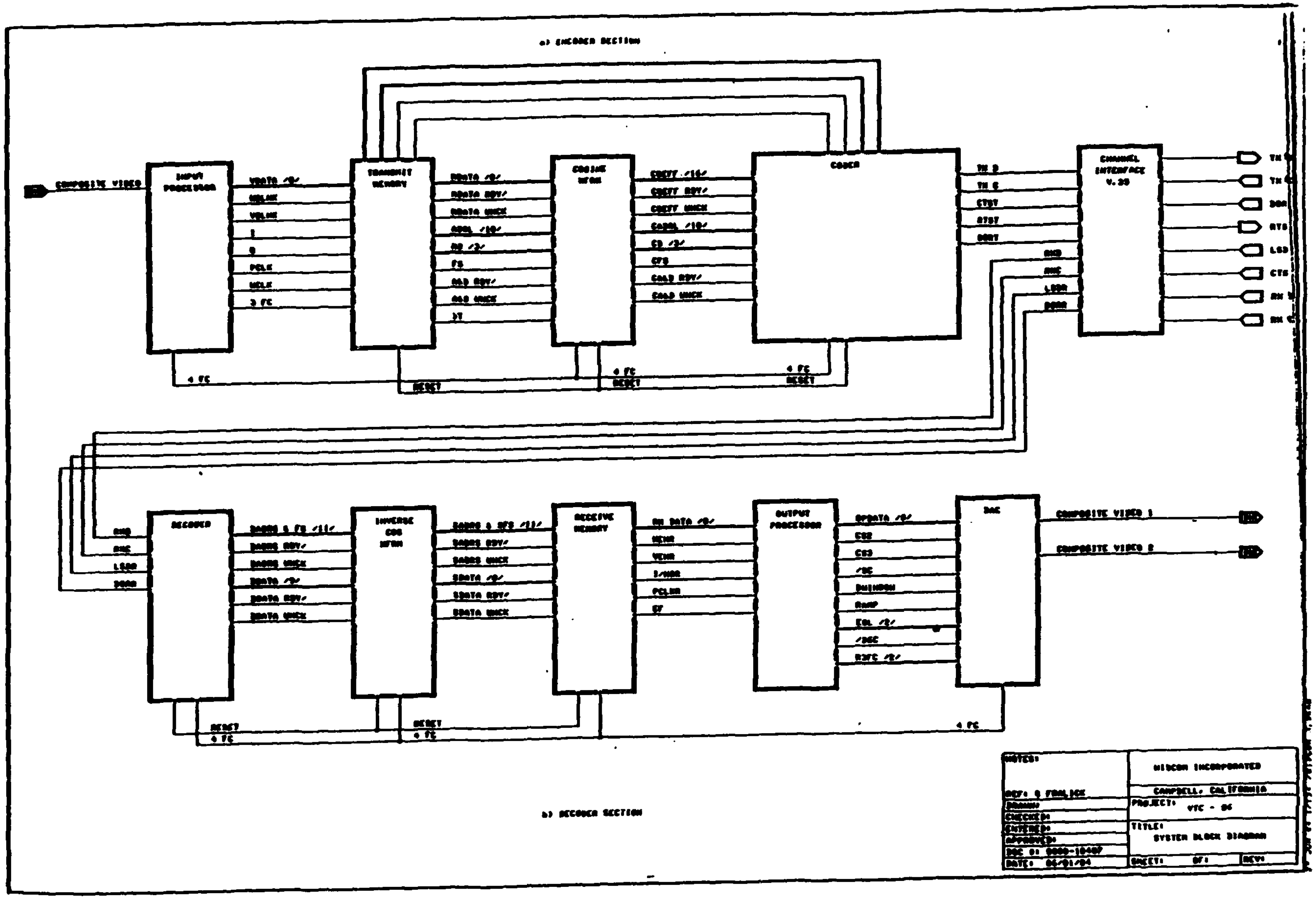
3.0 OPERATIONAL OVERVIEW

The blocks shown in Figure 3.0-1 represents the individual logic units which will occupy a single printed circuit card. Logic to perform several of the identifiable functions has, in some cases, been combined into a single card to simplify system fabrication.

3.0.1 ENCODER SECTION

The Input Processor Board contains circuitry to decompose the composite video into sync, luminance, and chrominance components. It also contains circuitry to convert the luma and chroma signals into a sequence of digital words. The sync signals are used to generate the system clocks and to synchronize the A/D converters. The chroma and luminance signals are A/D converted and passed as a sequence of parallel 8-bit bytes to the Transmit Memory Board. Horizontal and vertical blanking signals are also sent to identify the beginning of each line and field.

The Transmit Memory Board (TMB) consists of an IIR filter, two frame buffers and a Block Difference Processor. The TMB processes the sequence of 8-bit pixels performing a temporal average using the IIR filter. The resultant image is stored in the first frame buffer for later comparison with a reference image stored in the second frame buffer. The Block Difference Processor divides each of the two images into blocks of 8 rows of 8 pixels and performs a block by block comparison to find those blocks requiring replenishment. The Transmit Memory Board also accepts two data words from the Rate Buffer which specify the comparison threshold and the coding distortion for the current block. The threshold is used to decide whether the block has changed enough from the reference block to require replenishment. The distortion is a parameter that will be tagged to each block which exceeds the threshold and will be used later during the coding process. When a block is different enough from the reference image, it is transferred into the reference frame store and also passed to the Cosine Transform Board for further processing. The Transmit Memory Board also generates a frame sync signal when it starts to analyze the



SYSTEM BLOCK DIAGRAM

Figure 3.0-1

first block in a new frame, and counts runs of non-different blocks. These runs are sent to the Cosine Transform Board as the run-length address of the block being processed.

The Cosine Transform Board accepts the blocks as a sequence of 8-bit pixels with a distortion parameter and an address run length. The distortion and address are saved as tags. The CTB performs an 8-point Discrete Cosine Transform (DCT) on each of the rows, storing the results temporarily until an 8-row block is complete. The CTB then performs an 8-point DCT on each of the 8 columns in the block, again storing the results temporarily. The results of the DCT calculation are accurate to 16 bits; however, only 10 bits are required for coding. Therefore, the 64 elements of each block are sent to the coder as a sequence of 10 bit words. The CTB reorders the scanning sequence of the block elements so as to provide the diagonal scanning described in the algorithm report. The address and distortion tags are also sent to the coder.

The Coder Board consists of a quantizer, a state machine controlled coder, a set of lookup tables and an IIR filter. The distortion tag is used to establish the step size of the quantizer, which is then used to quantize all of the 64 DCT coefficients. The coder codes the value of distortion once every 4 blocks, the block run address for every block, and the 64 coefficients, all in accordance with the coding algorithm in Reference 1. The IIR filter is used to calculate the running mean of the coefficients to select the appropriate coding table. Coded results are passed to the Rate Buffer as 8-bit bytes. The Rate Buffer consists of an 8Kbit memory operated as a FIFO and an IIR filter. The coded data is written into the memory as it arrives from the coder, and read from the memory at a constant rate determined by the channel clock. The IIR filter keeps track of the smoothed value of the buffer fullness. This value is used to determine the current value of distortion and threshold parameters which are passed to the Transmit Memory Board.

The Channel Interface Board (CIB) provides a full duplex electrical interface to the communications channel. The channel is assumed to meet the CCITT V.35 specification. The CIB includes a set of buffers to convert the TTL level signals from the Rate Buffer to the balanced signals required to meet the V.35 interface spec. It also includes a set of circuits to receive the V.35 balanced signals and convert them into TTL levels.

3.0.2 DECODER SECTION

The Decoder Board consists of an input FIFO and frame synchronizer, a state machine controlled decoder, a set of look up tables and the circuitry to regenerate the digital values of the DCT coefficients. This latter may be called a "dequantizer"

since it performs the function complementary to quantization. The input FIFO provides an 8Kbit rate buffer between the constant input rate of the channel and the variable rate of the decoder. The frame synchronizer recognizes frame sync words, keeping the decoder in sync when uncorrected errors occur. The decoder operates on the input data one bit at a time to decode the values of the distortion, address run length, and DCT coefficients. The distortion is used to dequantize the coefficients. The resulting DCT coefficients are sent to the Inverse Cosine Transform Board (ICTB) as a sequence of 9 bit words. The address run length is sent as a single 10 bit word once per block. The beginning of a frame is also sent to the ICTB as a separate signal.

The Inverse Cosine Transform Board (ICTB) accepts the sequence of 64 coefficients and reorders them into an 8 x 8 block for processing. First the Inverse Discrete Cosine Transform (IDCT) of each row is calculated and stored temporarily, then the IDCT of each 8 element column is calculated. The resulting block is rescanned and sent to the Receive Memory Board. The frame sync signal and the address run length of each block are also passed to the Receive Memory Board.

The Receive Memory Board consists of two frame buffers and a digital IIR filter. The address of each block is used to write the decoded pixels of that block in the appropriate location in the reference frame store. Except for coding distortion and channel errors, this frame buffer will contain a replica of the transmit reference frame buffer, delayed by the coding/decoding process. The IIR filter serves several purposes. During even fields, the filter operates to interpolate temporally between received frames by averaging the two frame buffers and using the result both as an output to the Output Processor Board and as an input to the refresh frame buffer. During odd fields, the filter operates to interpolate between vertical lines by averaging two lines from the refresh buffer to regenerate the required 480 lines from 240 lines transmitted and stored. During blanking intervals, the filter provides both temporal and spatial interpolation of the color signals. The interpolated data is sent to the Output Processor Board as a sequence of 8-bit bytes transferred under control of the Output Processor Board (OPB) signals (pixel clock, horizontal blank, vertical blank, and color request).

The Output Processor Board contains circuitry to perform horizontal interpolation, generate the horizontal and vertical sync signals, remodulate the color carrier, and combine the components into a digital representation of the composite video signal.

The DAC board contains a D/A converter to generate an analog composite video signal. This signal is filtered and amplified to provide a standard NTSC signal. The Output

Processor Board is capable of being genlocked to an external source of composite sync.

3.0.3 DATA FLOW AND INTERFACE TIMING

Data flows from the video source into the input frame buffer synchronously in raster format at 60 fields per second. The chroma data is stored at the bottom of the raster. The input frame store contains 232 lines of luma followed by 8 lines of Q and 16 lines of I information. This results in a 256 x 256 frame (see the Transmit Memory Board memory map). All subsequent processing is based on 8 x 8 blocks of pixels selected from this frame; hence, all subsequent data flow is in 64-pixel blocks until the Receive Memory Board once again stores a 256 x 256 frame in format identical to the input frame store. The Output Processor Board converts this frame to the 60 field per second raster of NTSC specification.

The system has been parsed into logic blocks which result in a simple data flow between elements. Except for the Input and Output Processor Boards, all data is transferred in 64 pixel blocks. Each block includes its own address as a run length from the last block. In the encoder, the coding distortion is also transferred with each block. The first block in the frame is always marked with a frame sync bit. This allows each processor in the system to operate asynchronously—such operation being required to maintain low peak processing rates.

Data is transferred synchronously between the Input Processor Board and Transmit Memory Board and between the Receive Memory Board and the Output Processor Board. All other data transfers are asynchronous and require a two signal handshake between the logic elements involved. The sending element puts valid data on the data lines and raises a "ready" signal. The receiving element generates an "unload clock" when it has taken the data. The data must be valid before the "ready" signal is activated. The data must remain valid until the "unload clock" returns to its inactive state. Figure 1-2 shows this generic handshake.

APPENDIX C

ILEX VOPAC VOICE DIGITIZER USER MANUAL

ilex

USER MANUAL

VOPAC VOICE DIGITIZER

July 1984

Ilex Systems, Inc.
1423 South Milpitas Blvd.
Milpitas, California 95035

[redacted]

TABLE OF CONTENTS

TABLE OF CONTENTS (Continued)

ILLUSTRATIONS

TELEPHONY--BACKGROUND ON TIE-TRUNK CONFIGURATION

All input/output audio and signaling connections are made to the 50-pin phone line connector on the back of the unit. The connector provides the balanced 600 ohm termination for the phone lines as well as the required contact closures and sensors for signaling purposes.

The signaling consists of on/off hook and dialing information. These signals are passed between the signaling circuit (Vopac) and the trunk circuit (PBX) by means of relay contact closures. When a contact is closed, a current flows in a loop which is sensed by, for example, an inductive sensor. The following is a brief summary of the sequence used in establishing a connection between a voice digitizer and a PBX.

It will initially be assumed that the Vopac is originating the calling sequence toward the PBX. The Vopac does this by sending an "off hook" signal to the PBX. An off hook signal is a contact closure completing a circuit. The PBX senses this closed loop and responds by sending its off hook signal toward the Vopac in the form of a relay contact closure. This contact closure by the PBX is in turn sensed by the Vopac, and the connection is then completed. Rotary, or pulse, dialing is simply a series of on/off hook pulses passed on the E lead after the original connection has been established. DTMF, or Touch-Tone dialing consists of groupings of tones passed over the audio lines.

The circuits carrying the on/off hook information are provided for by what is known as E & M signaling. The E lead (or E circuit) provides the contact closure that indicates to the trunk that the signaling circuit (Vopac) is off hook. The M lead (or M circuit) provides the contact closure that indicates to the Vopac that the PBX is off hook. Variations in the wiring of E and M circuits are used in different applications, and will now be discussed.

Differences in specific wiring applications have led to E&M signaling Type I and Type II. The configurations for these are shown in simplified form in figure 4. The Type I interface is the historic 2-wire configuration and it is the preferred interface for electromechanical systems.

The Type II arrangement was developed primarily for use with electronic switching environments. It is a 4-wire, fully looped arrangement in which open and closure signals are used in each direction. An on-hook signal is designated as an open circuit, while an off-hook signal is a closed circuit. The standard Type II interface provides closure to -48 volt on the M-leads and closure to ground on the E-leads. The Type II interface has the advantage that trunk circuits can be directly connected together without interposing converter circuits or signaling circuits. This is known as a back-to-back connection.

GENERAL

If the Vopac is to be rack mounted, the rubber bumpers on the chassis bottom must be removed. The unit may now be installed in the rack using the slide mounts.

There are three types of electrical connections required for the Vopac. These are power, telephony and digital.

POWER CONNECTIONS

The Vopac is equipped with a power connector that incorporates a fuse and a voltage selector. Allowable voltage settings are 100, 120, 220, and 240 Volts AC. Before plugging the unit in, the voltage setting should be checked. The voltage setting can be viewed on the card beneath the fuse through the clear plastic window on the AC line plug. If the value is not correct, proceed as follows:

1) Slide the clear plastic door to the left. The AC line cord must be disconnected from the socket on the back of the Vopac (see figure 3).

2) Rotate the fuse pull to the left, removing the fuse.

3) Using a small screwdriver, or other small prying instrument, remove small PC board from under fuse holder.

4) Select operating voltage by opienting the PC board so that the desired voltage is on the top left side. Push board firmly into module slot.

5) Rotate fuse-pull back into normal position and re-insert fuse into holders, using caution to select correct fuse value (see table __ for correct fuse values).

With the Vopac set for the correct input voltage and with the correct fuse in place, plug the AC line cord firmly into the AC socket in the back of the unit and into the wall outlet.

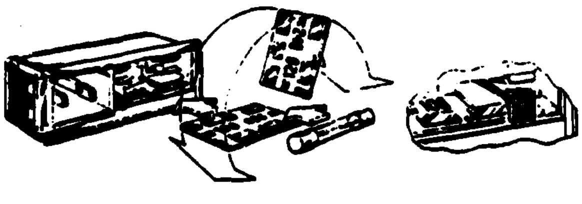


Figure 3 -- Voltage Selection

100V -- 3 A	220V -- 1.5 A
120V -- 3 A	240V -- 1.5 A

Table 3 -- Fuse Ratings

Chapter 2:

INSTALLATION

SPECIFICATIONS

Power Requirements

100, 120, 220, or 240 Volts AC
47 - 63 Hz; 120 Watts

Environment

0% - 90% Relative humidity, non-condensing

0 - 50 degrees Celcius

Physical

19 inch rack mounting

17 inches wide, not including slide mounts

3.50 inches high

20 inches deep.

Weight: 40 lbs.

Connections

Telephony connections are made to a 50-pin ribbon type connector. Telephony connections support 2- or 4-wire audio terminating to 600 ohms. Supported signaling types are E & M Types I and II and back-to-back connections.

Digital connections are made to a 25-pin D-type connector. MIL-STD-188C or RS-232C specifications are strap selectable.

An RJ11 minimodular jack is provided for handset plug-in.

Power connections are made to a CEE-22 power connector on the rear of the unit.

Data Format

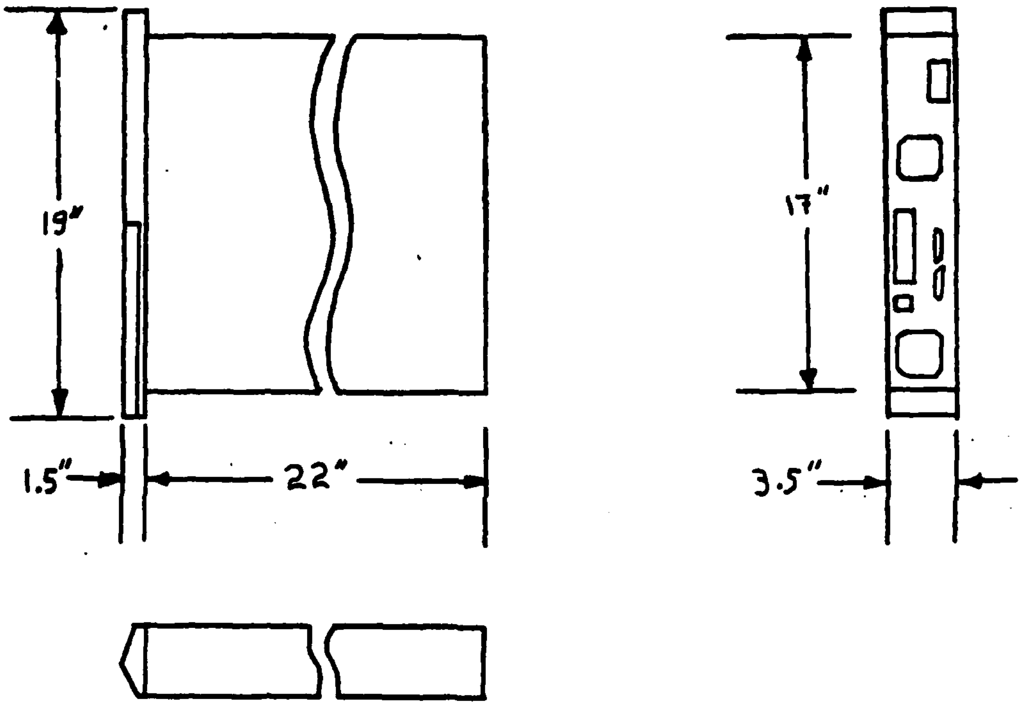
Bit rates of 2400 and 4800 bps are standard. LPC10/43 and DBSS data formats are selectable at 2400 bps. DBSS is the 4800 data format.

PIN NO.	SIGNAL
1, 26	TT, TR
2, 27	RT, RR
3, 28	BAL1, BAL2
9, 34	DG
35	PLOOP
11, 36	MP, MN
12,	ENR
13, 38	+5 Vdc
14, 39	+15 Vdc
15, 40	-15 Vdc
17, 42	EP, EN
19, 44	BATT GND
20, 45	BATT
21, 46	SP, SN
25, 50	RDETH, RDETL

TABLE 1: PIN DESIGNATIONS FOR
50-PIN PHONE LINE CONNECTOR

PIN	SIGNAL	TYPE
1	Protective ground	
2	Transmitted data	Output
3	Received data	Input
4	Request to send	Output
5	Clear to send	Output
7	Signal ground	
8	Carrier detect	Input
15	External clock	Input
17	Receive clock	Input
20	Data terminal ready	Output
24	Transmit clock	Output

TABLE 2: PIN DESIGNATIONS FOR
"MODEM" RS-232 CONNECTOR



dimensions are in inches

Figure 2 — Vopac outline

PLOOP, MN - Detects loop current of 3 mA or more. A low impedance version of MP, MN used for extension line connections.

SP, SN - A spare output of contact closure towards the PBX.

EP, ENR - Full wave rectified form of ring signal taken from RDETH, RDETL.

DG - Vopac logic ground.

+5VDC - +5 volt direct current power supply output through a 100 ohm resistor from the digitizer.

+15VDC - +15 volt direct current power supply output through a 100 ohm resistor from the digitizer.

-15VDC - -15 volt direct current power supply output through a 100 ohm resistor from the digitizer.

PHYSICAL DESCRIPTION

GENERAL
The Vopac is designed to be operated on a desk top or in a standard 19-inch rack. The unit is shipped with rubber bumpers installed for desk top use. These must be removed for rack mounting.

The Vopac is 17-inches wide without slide rack mounts, and requires 3.5 inches of vertical space. The unit is 22 inches deep (not including the front panel). All outer physical dimensions are given in figure 2.

CONNECTORS
Pin designations for the 50-pin telephone connector are given in table 1, and those for the 25-pin modem connector are given in table 2. A brief description of the signals appearing on the telephone connector follows:

Audio signals:

- TT,TR - 600 ohm, transformer coupled input to Vopac. TT is the "tip" line and is the more positive (ground). TR is the "ring" line and is the more negative.
- RT,RR - 600 ohm, transformer coupled output from Vopac. RT is the "tip" line and is the more positive. RR is the "ring" line and is the more negative.
- BAL1, BAL2 - Balance winding terminals used in a 2-wire audio configuration.

Signaling:

- EP, EN - Contact closure toward PBX from Vopac. The more positive voltage must be on the EP terminal. The closure will handle 125 mA, with 300 volt max. open circuit voltage.
- MP, MN - Circuit provides detection of contact closure from the PBX. A closure is detected when 15 or more volts appears across the terminals. The more positive voltage must be on the MP terminal.
- BATT GND - Phone system ground.
- BATT - Regulated -48 volt, 125 mA supply provided by Vopac.
- RDETL,RDETH - Ring detect low and high. Detects ringing voltage of 30 VRMS min., 15 Hz min.

APPLICATIONS

The voice digitizer is designed to be used in two ways: as a communicating device and as a conversion element for digital voice storage and retrieval systems. As a communications system, the Vopac provides a digital bit stream that can be encrypted to form a secure communication link. A digital bit stream of 2400 or 4800 bps can also be used to expand the use of a data line. On a line that is capable of handling 9600 bps, up to four separate voice communications can be sent using the 2400 mode with separate sets of units. Voice conversation can also share the line with other transmitted digital data, eliminating the need for separate voice and data networks. A typical transmission application is shown in figure 1. In voice storage applications, a 20 megabyte disc can store over eight hours of continuous speech at 4800 bps.

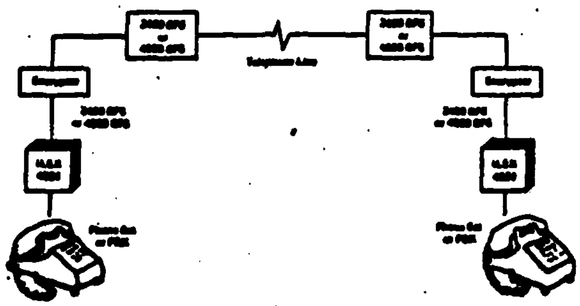


Figure 1 -- Basic secure voice transmission application

The LPC10/43 algorithm allows for hookup with any other system type using this algorithm. The DBSS algorithm is a proprietary algorithm and it offers the advantage of a more natural sounding voice quality. The DBSS algorithm particularly enhances the fullness of the tonal qualities, making individual voices more recognizable. DBSS also provides for telephony signaling within the data frames.

The data frame format of DBSS provides digital data to model the human voice. The frames carrying information for the voiced sounds (vowels, consonants such as "m" or "g") provide information on sound, power and pitch. The unvoiced sounds consist mainly of "white" noise signals, and leave room in the data frames for passing extra telephony and sync signals. This is where synchronizing signals between the two units are passed, as well as on/off hook and dialing information.

VOPAC/OPERATOR INTERACTION

The operator controls the Vopac through the front panel. The front panel configuration is "soft", meaning that all keys are scanned by Vopac software, and required action is taken strictly under software control. The important settings are saved in non-volatile memory in the event of a power-down or brownout.

All mode selection and level setting is done through the front panel. On installation, levels for input, output, and echo return loss are set in accordance with the interfaced equipment. Front panel control is done using the 16-character alphanumeric display and the pushbuttons. The display tells the operator the Vopac operating mode. It also gives a numerical reading of various levels.

Located below the alphanumeric display is a bar graph display that gives peak reading levels of audio inputs and outputs. The alpha-numeric display, when left in operating mode, can be used to monitor various levels, or, more commonly, it can be used to monitor line status. It can also reaffirm to the operator that the correct number is being dialed.

SELF TEST

In addition to mode selection and level adjustments, a self test can be made from the front panel. The self test steps through the Vopac memory circuits, the high speed numerical processor (number cruncher), and the analog circuitry. As each individual test is completed, the result "OK" or "FAIL" is indicated on the display. At the conclusion of the test sequence a summary will be displayed indicating that all tests were passed or, if not, which board within the unit is bad.

Another test mode that is incorporated within the Vopac is a loopback mode. This is used to determine the functionality of the local Vopac unit by returning the local outgoing bit stream directly to the local input. The integrity of the digital link and the functionality of the remote unit are also determined, as data is looped back from the digital input as well.

of 10 pps is used in the United States, while 20 pps is used in some overseas telephone networks.

A handset may be connected to the unit via an RJ11 connector on the rear panel. The handset provides for single instrument voice communication without dial capability.

A Watson Phone may be connected to the back of the unit. It provides single instrument voice communications with dial capability and ring detection.

When connected to a system that provides an external clock, the Vopac squelches the analog output and prevents additional output to the earpiece when there is no incoming digital signal.

ANALOG--Audio Level Control

Following installation, the audio levels of both incoming and outgoing signals are set using the front panel controls. During operation, an Automatic Level Control (ALC) system is available to monitor calls on an individual basis. The ALC system, when enabled, readjusts input and output levels to compensate for loudness differences due to individual voices, handset holding habits and for the variations in levels if access to the public telephone network is made. The ALC system operates on a call-by-call basis, and the levels are returned to the front panel settings at the end of each call.

Another level control feature is the FCC Limiter. When enabled, the outgoing audio levels are held to within the regulatory demands of the FCC. This ensures that the Vopac can be used with any application involving the public telephone network.

In addition to level controls, the Vopac provides an echo cancel feature for use with 2-wire audio circuit installations.

DIGITAL

The Vopac is designed to operate in either full- or half-duplex mode across the digital data link. The digital link connection is provided through a rear panel, 25-pin male D-subminiature connector. The connector is wired in accordance with RS-232C. The Vopac is normally interfaced to a modem or other digital data link port via this connector. Signal levels are jumper selectable for either RS-232 or MIL-STD-188C specifications.

The transmit clock source is user selectable. The source selected is based upon the installation configuration. It is external, when, for example, the clock is supplied by the modem. It is internal or derived from the receive clock if the data link port expects the transmit clock from the Vopac.

ALGORITHMS

The Vopac provides two digital voice algorithms: the LPC10/43 (FED STD 1015) algorithm at 2400 bps, and the DBSS proprietary algorithm at both 2400 and 4800 bps.

FUNCTIONAL DESCRIPTION

GENERAL

The VOPAC Voice Digitizer is a state-of-the-art voice digitization unit that uses advanced algorithms to analize analog speech and voice signals and transform them into a set of digital parameters. This digital data can then be transmitted over a digital data link operating at 2400 or 4800 bits per second. When this data is sent to another Vopac unit, the digital data is synthesized back to analog, and speech is regenerated. The Vopac performs both the analysis and synthesis functions independently.

The Vopac is supplied with both the ILEX proprietary DBSS algorithm and the LPC10/43 algorithm. The DBSS algorithm supports transmission of telephony signaling.

ANALOG--Input Flexibility

The Vopac audio input/output interface is designed for considerable flexibility. The audio input can be a telephone system such as a PBX, a direct microphone, or a tape recorder. An input is also provided for a handset directly, although this is used primarily for testing purposes.

ANALOG--Telephone Interface

The telephone interface provides for connection to PBX's, the ILEX Watson (ringdown) Phone or other equipment having standard telephone interfaces. The Watson Phone is the ringdown telephone that operates with the Vopac. The PBX interface allows for 600 ohm, 2- or 4-wire audio circuits. Signaling can be set up for Type I and Type II E & M signaling, and various USOC interfaces. Signaling is compatible with connection to a tie-trunk, an extension line, or a Watson Phone.

The tie-trunk interface supports both 2-wire and 4-wire audio operation. The Vopac allows both inward and outward dialing, and supports through-dialing to the remote Vopac to access any telephone instrument within the far-end PBX or dial network.

With the PBX extension interface Vopac operates as a 2-wire extension station within a PBX. The Vopac will detect ringing signals, and answer the extension when dialed. It will also support dialing through the PBX to the remote Vopac and its associated telephone network.

The Vopac unit can be used with either rotary or Touch-Tone (DTMF) dialing inputs. The Vopac will output dialing information as rotary pulses or as DTMF signals. Dialing into the unit can always be done with either rotary or DTMF. The Vopac software recognizes voice signals, and ignores DTMF when voice signals are present. This prevents the unit from misinterpreting voice signals as DTMF. Dialing can otherwise be done at any time.

For rotary dialing information, rates of either 10 or 20 pulses per second (pps) can be selected by the operator. The dial rate

Chapter 1:

GENERAL INFORMATION

INTRODUCTION

This manual provides information that will enable the user to install and set up the VOPAC Voice Digitizer for proper, continued operation.

A background is given of the various interfaces provided by the Vopac, including the analog, digital, and human interactions. A background of the telephony connections that will be encountered on connection to a PBX is given in order to supply the user with a brief understanding of the required hook-up configurations.

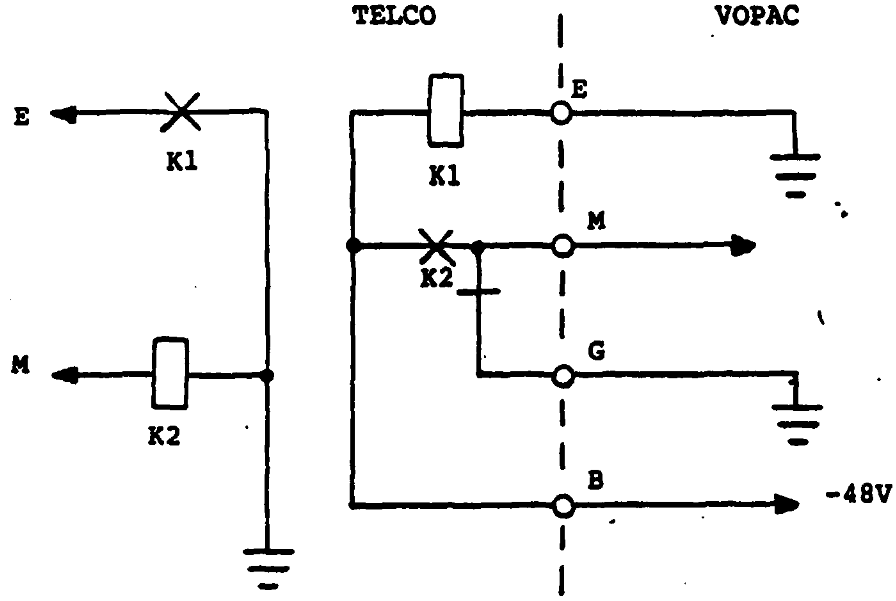
Upon installation, the user is stepped through adjustments necessary to set proper audio levels and functions in order to provide trouble-free performance by the VOPAC Voice Digitizer.

TELEPHONY CONNECTIONS--AUDIO HOOK-UP

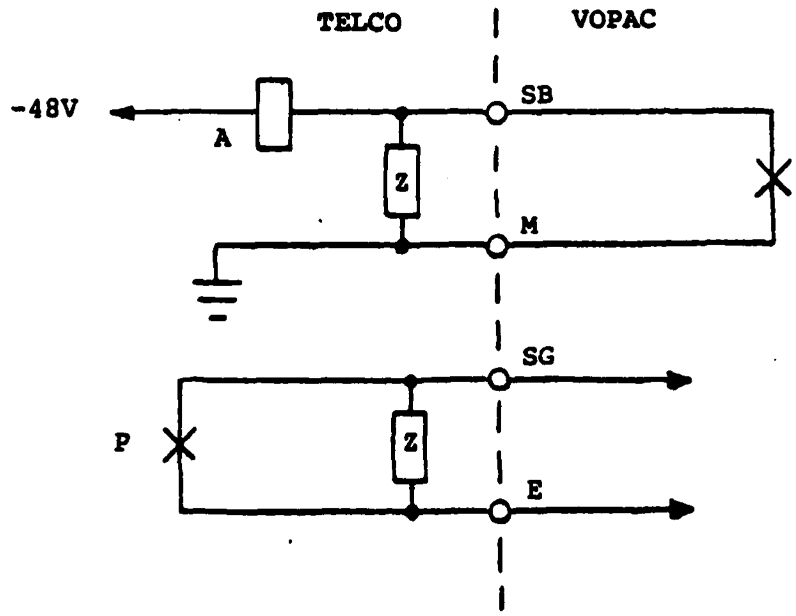
A tie trunk interface should be 4-wire if possible. The audio wires at the PBX will likely be labeled T, R, and T1, R1. The "T" and "R" stand for "tip" and "ring", and refer to the parts of the phone jack that would be used if switching were done manually by an operator. The T/R pair are for audio signals generated by the Vopac and going toward the PBX and the T1/R1 pair are for audio coming from PBX towards Vopac. If the PBX interface is 2-wire, the wires should be labeled T and R. There should be no measurable voltage across any T/R pair.

The corresponding wires on the Vopac are labeled RT, RR, TT and TR. The RT/RR pair are the signals going from the Vopac towards the PBX and the TT/RR pair are the signals coming toward the Vopac from the PBX. For a four-wire interface, connect T to RT, R to RR, and T1 to TT, R1 to TR.

For a two-wire interface, connect RR from the Vopac to R on the PBX. Connect TT on the Vopac to T on the PBX. TR and RT on the Vopac must also be directly connected to each other. A balance network must also be installed between BAL1 and BAL2 on the Vopac.



(a) Type I



(b) Type II

Figure 4 - E&M Signaling

TELEPHONY CONNECTIONS--SIGNALLING HOOK-UP

The E & M leads will be referred to as "contact closure leads", since they provide the circuit through which a closure to ground or to -48V is detected. The markings on the leads from the Vopac will be explained first, as the corresponding line at the PBX may be less well defined.

EP and EN provide the circuit that indicates a contact closure from the Vopac towards the PBX (E circuit). The positive-most voltage is applied to the EP lead. MP and MN provide the circuit that senses a contact closure from the PBX towards the Vopac (M circuit). The more positive voltage is applied to the MP lead.

The signalling hook-up to various Facility Interface codes are given in Table 4. The different Facility Interfaces described are as follows:

31-TT-EG - Standard Type I E & M Interface
11-TT-EG - Standard Type I E & M Interface
12-TT-MB - Standard Type II E & M Interface.

The connections for installation to an extension station line are also given in Table 4.

The Watson ringdown phone is a unit complete with cable and a 50-pin connector. To install the Watson phone, simply plug the cable into the 50-pin connector on the back of the Vopac.

DIGITAL

All connections to the digital interface are made through the 25-pin "modem" connector. The pin designations are given in Table 2.

To connect the Vopac to the digital communications link, simply plug the cable from the modem, encryptor, or other data handling unit into the "modem" connector on the rear of the Vopac.

Bell System Facility
Interface

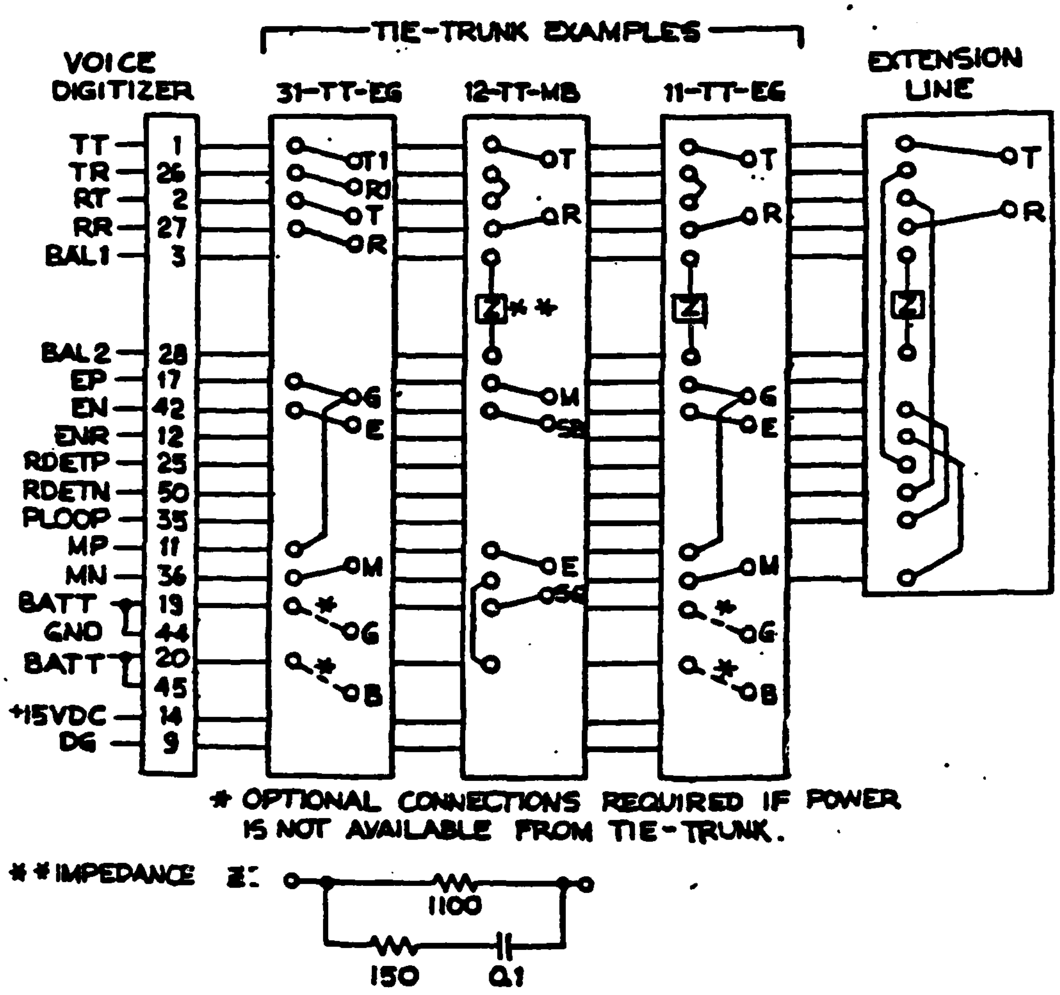


TABLE 4: TIE TRUNK
CONNECTIONS

Chapter 3:

OPERATION

FRONT PANEL USAGE

INTRODUCTION

On the front of the Vopac there are 10 pushbuttons marked as shown in figure 5, a power switch, an alphanumeric display and a light bar display. Figure 5 is shown on page 24 and is described in the following paragraph.

The functions controlled through the front panel are set up in "menus" associated with each of the seven left-most pushbuttons. The operator moves to different functions within each menu using the SKIP pushbutton. These functions are preceded by an S in the table. The function name selected appears on the alphanumeric display, and the status of the function is also shown. The function options are set with the ↓ and ↑ pushbuttons. These are also indicated in the table. A solid red light appears within each menu pushbutton when it is selected, and flashing lights appear within the ↓ ↑ and SKIP buttons when these are operative.

This manual will step through the functions associated with checking the operation of the signalling leads and setting the audio levels that need to be set before normal service can begin. A complete listing of all functions and their uses is given at the end of this chapter.

SELF TEST

Turn the unit on. After a few seconds the unit will beep and the display will read RESET.

Push the TEST button. A red light appears within the TEST button indicating that it is in use. The display reads SELF TEST: OFF and flashing lights appear within the ↓ ↑ and SKIP buttons, indicating that these are now active. Pressing the SKIP button changes the display to indicate the other functions under the TEST mode. These are TEST TONE: XXX and 2400 MODE: XXX. Return to SELF TEST. Pressing either the ↓ or the ↑ buttons causes the display to read SELF TEST: ON. The Vopac then steps through its self test, indicating all sections "OK", and finally read "ALL TESTS PASSED" before ending the test. If all tests are not passed, see Troubleshooting Chapter. Pressing the TEST button again causes the light to go off within the TEST button and the display to read IDLE. The lights above the ↓ ↑ and SKIP buttons also go off.

OPERATIONAL VERIFICATION AND LEVEL SETTING

GENERAL
At this point the signalling leads and audio leads have been connected to the appropriate telephone system so that operation of the signaling leads (contact closures) should be verified and audio levels set. The operator should be briefly familiar with the use of the ↓ ↑ and SKIP pushbuttons after completing the self test run.

SIGNALING CHECK
On the front of the Vopac press the DISPLY button. A red light appears on the DISPLY button and DISPLAY: XXX appears on the alphanumeric display. Use the ↓ and ↑ buttons to select DISPLAY: CALLS. Press DISPLY again to exit the mode. The alphanumeric display reads IDLE and all lights go off over the pushbuttons.

From a nearby telephone that is connected to the PBX dial the tie-trunk number that the Vopac has been wired to. The display now reads OUTGOING CALL. Dial some digits. The display indicates the digits dialed. This completes outgoing signal verification. Leave the phone off-hook for the next step.

INCOMING AUDIO LEVELS
Press the DISPLY button again. Lights appear above the buttons again and DISPLAY: CALLS appears on the alphanumeric display. Press SKIP. This changes the display to read BAR: XXX. Use the ↓ and ↑ buttons to select BAR: INPUT LEVEL. Press the AUDIO button. Use SKIP to select ALC SYSTEM: XXX and ↓ or ↑ to select ALC SYSTEM: OFF. Now use SKIP to select INPUT: XXX. Use ↓ or ↑ to select INPUT: PBX. Push the SKIP button to display FROM PBX: XXX. The number, in decibels (DB), that is displayed is the number that the machine was set to the last time the level was set. The ↓ and ↑ pushbuttons lower or raise the value in 3 DB increments to limits of 0 DB and 45 DB. Select FROM PBX: 0 DB. Use the SKIP button to select TO PBX: XXX. Select TO PBX: 0 DB (using ↓). Use SKIP to return to FROM PBX: 0 DB.

With the nearby telephone that was used to dial the tie-trunk number, talk in a normal voice into the handset. Use the ↓ and ↑ pushbuttons to set the level so that the peak of the bar lights (indicated by the lingering light) is about one-half to three-quarters scale. If a Touch-Tone telephone is available, the input level may be set by adjusting FROM PBX for about half scale while pressing and holding any digit. Push the DISPLY button again to turn this function off.

OUTGOING SIGNAL LEVEL
Press TEST and, using SKIP, select TEST TONE: XXX. Use ↓ or ↑ to select TEST TONE: ON. Dial the tie line number from the nearby telephone as before to access the Vopac. Press

AUDIO. SKIP to TO PBX: 0 DB. While listening on the telephone handset, adjust the output level using ↑ until the tone is approximately as loud as local dial tone -- comfortable, but not too loud.

This completes outgoing signal level setting. Press TEST and turn off the TEST TONE. Press TEST again to return the Vopac to IDLE.

ECHO RETURN LOSS

Dial the tie-trunk number from a quiet nearby phone that is connected to the PBX. Press TEST, and select TEST TONE: ON. Press DISPLY. SKIP to DISPLAY: XXX and use ↓ or ↑ to select DISPLAY: ECHO RET. Press DISPLY again to exit. The display will indicate the echo return and it should be less than (more negative than) -6 DB. If not, and a four-wire installation is being used, then the PBX provider should be called. If a two-wire installation is used, turning down either the input or output audio levels will improve the echo situation. Turn them each down a step at a time until ECHO RETURN is below -6. Select TEST TONE: OFF when level setting is complete.

ECHO CANCELLOR

Press PBX. SKIP to ECHO CANCEL: XXX and use ↓ or ↑ to set ECHO CANCEL: ON.

AUTOMATIC LEVEL CONTROL (ALC)

It is recommended that the ALC system be enabled for normal use to compensate for any loudness differences that may occur. Both the "from" and "to" PBX levels are adjusted automatically and in a slow manner that is imperceptable to most users. The ALC system is enabled by pressing AUDIO, and SKIPping to ALC SYSTEM: XXX. ↓ and ↑ will turn the ALC system ON and OFF.

Generally, any changes to the front-panel level settings (which are used as a starting point for the ALC system) should be made with the ALC System OFF so that the adjuster is not confused by the ALC contribution to the levels being observed.

VOICE QUALITY CHECK

The voice quality produced by the Vopac can be checked using the loopback mode and the handset plugged into the rear of the unit. Press the CHANL button and SKIP to LOOPBACK: XXX. Use ↓ or ↑ to select LOOPBACK: IN. Press AUDIO, and SKIP to INPUT: XXX. Select INPUT: HANDSET. Now speak a few words or phrases into the handset. Your voice should be heard in the handset earpiece after a delay of approximately 200 msec. If the DISPLY button is used to select BAR: INPUT LEVEL, the bar graph will indicate the peak input level of your voice.

When speaking into the handset, your own voice should sound much as it does on a regular telephone, with no pops, clicks, dropouts or unusual distortion.

The voice quality transmitted using the different modes of operation may be checked. Press the CHANL button. Use SKIP to select BAUD RATE: XXX. The user can now choose either the 2400 or the 4800 rate with the ↓ or ↑ pushbuttons. By pressing the TEST button, and SKIPing to 2400 MODE: XXX, the operator can select either the DBSS or the LPC10 algorithm to be used for speech analysis/synthesis. At the 4800 baud rate, only the DBSS algorithm is available.

A full description of all of the functions associated with each button is listed in the following section.

TEST | TTY | CHANL | PBX | AUDIO | DISPLY | INFO | ↓ | ↑ | SKIP

TEST

S Self Test: Off
: On

S Test Tone: Off
: On

S 2400 Mode:
: LPC10
: DBSS

TTY

Not Implemented

CHANL

S Baud Rate:
: 4800 bps
: 2400 bps

S Loopback: Out
: In

S TX Clock:
: Internal
: RX Clock
: External

PBX

S Echo Cancellor: On
: Off

S Output:
: Rotary
: DTMF

S Rotary:
: 10 pps
: 20 pps

S PBX:
: Tietrunk
: Extension
: Ringdown

AUDIO

S Input: PBX
: Handset

S From PBX: xx db
lowers value
raises value

S To PBX: xx db
lowers value
raises value

S ALC System: On
: Off

INFO

Software Version Number

↓ / ↑

S Display: Calls
: Echo Ret
: ALC Gain

S Bar: In & Out Level
: Input Level
: Noise Floor

Figure 5 -- Pushbutton Designations

PUSHBUTTON FUNCTIONS -- A COMPLETE DESCRIPTION

** NOTE ** SKIP indicates how to access the different functions after the listed button has been pushed, and the arrows (↓ ↑) show the implementation of the functions.

TEST -This pushbutton controls the test functions of the Vopac. It also provides for selection of the 2400 algorithm.

SKIP - SEFT TEST: OFF

↓ ↑ SELF TEST: ON - The Vopac will step through and test individual sections of the unit. Upon completion of each portion of the self-test, an indication "OK" or "FAIL" will appear following the section name. The display will appear as follows ("/" denotes "or"):

ANALOG	OK/FAIL
68000 PROM	OK/FAIL
68000 RAM	OK/FAIL
SHADOW RAM	OK/FAIL
COMMON RAM	OK/FAIL
CRUNCHER	OK/FAIL

At the conclusion of the test sequence a summary will be displayed indicating "ALL TESTS PASSED" if all sections are OK. "LOWER BOARD BAD" or "UPPER BOARD BAD" is displayed to indicate where the failed section(s) are located.

SKIP - TEST TONE: OFF

↓ ↑ TEST TONE: ON - The use of TEST TONE has been described in the first part of this chapter.

SKIP - 2400 MODE: XXX

↓ ↑ 2400 MODE: DBSS

↓ ↑ 2400 MODE: LPC10
This function selects the 2400 operating mode. The LPC10 algorithm, or LPC10/43, is a federal standard (FED STD 1015) coding and format and may be used in conjunction with any other device that provides this coding. The DBSS is a proprietary algorithm that provides a better voice quality with fuller, more accurate voice tone.

TTY - This function is not implemented.

CHANL - This pushbutton controls the characteristics of the primary digital channel on the Vopac.

SKIP - BAUD RATE: XXX

↓ ↑ BAUD RATE: 4800

↓ ↑ BAUD RATE: 2400
A baud rate of either 4800 bps or 2400 bps may be selected. The 4800 mode operates only in the DBSS algorithm while the 2400 mode provides for selection between DBSS and LPC10.

SKIP - LOOPBACK: XXX

↓ ↑ LOOPBACK: OUT

↓ ↑ LOOPBACK: IN - The loopback function loops back the local outgoing digital bit stream to the local input. It also loops back the incoming remote bit stream to the outgoing (remote) bit stream. The loopback function, as demonstrated in the voice quality check, can be used to determine the functionality of the local Vopac. By looping back data to the remote Vopac, the integrity of the digital link can also be determined. The loopback function forces the selection of TX CLOCK: INTERNAL.

SKIP - TX CLOCK: XXX - The clock source for the synchronizing clock on the transmit line from the Vopac can be selected.

↓ ↑ TX CLOCK: INTERNAL - The Vopac provides an internas clock source for the transmit line.

↓ ↑ TX CLOCK: RX CLOCK - The transmit line clock is derived from the clock on the receive line.

↓ ↑ TX CLOCK: EXTERNAL - The transmit line is run based on a clock outside the Vopac - as when supplied by a modem, for example.

PBX - PBX and telephone line related functions are selected through this pushbutton.

SKIP - PBX: XXX - This function selects the way that signaling protocols are handled. The appropriate selection is made depending on the physical interface to the phone line. These interfaces may be:

↓↑ PBX: EXTENSION - When the Vopac is connected to an extension station line. When using an extension line configuration, the "*" button on the keypad of a Touch-Tone phone must be used at the end of a call to ensure that the line connection between the units is taken down. This is recognized by software as an indication to "hang-up".

↓↑ PBX: TIE TRUNK - This is the preferred or normal selection for a full tie trunk hook-up.

↓↑ PBX: RINGDOWN - This is selected when the Watson ringdown phone is used.

SKIP - ECHO CANCEL: ON

↓↑ ECHO CANCEL: OFF
The echo canceller is used on telephone audio connections. The echo canceller is primarily for use with two-wire audio circuits. The canceller is automatically disabled if the local handset is selected.

SKIP - OUTPUT: DTMF

↓↑ OUTPUT: ROTARY
This function will determine whether the Vopac will output the dialing information as rotary pulses or as Dual Tone Multi-Frequency signals. The Vopac will accept as input both types of dialing information at any time.

SKIP - ROTARY: 10 PPS

↓↑ ROTARY: 20 PPS
This function selects either 10 or 20 pulses per second as the rotary dial pulse rate. Both the output and the input pulse rates are set with this function. Select 10 PPS for most PBX's in the United States.

AUDIO - Input and output audio levels and choice of inputs are provided by this pushbutton.

SKIP - FROM PBX: XXX DB - This function adjusts the audio signal level input to the Vopac on the phone line (PBX). The total range covered is from 0 DB to 45 DB. The number displayed does not include ALC or FCC limiter effects.

↓ lowers the value in 3 DB steps.

↑ raises the value in 3 DB steps.

SKIP - TO PBX: XXX DB - This function adjusts the audio signal level output by the Vopac on the phone line (PBX). The total range covered is from 0 DB to 45 DB. The number displayed does not include ALC or FCC Limiter effects.

↓ lowers the value in 3 DB steps.

↑ raises the value in 3 DB steps.

SKIP - ALC SYSTEM: ON

↓ ↑ ALC SYSTEM: OFF
The Automatic Level Control readjusts the input and output levels on a call by call basis to account for loudness differences encountered during use.

SKIP - FCC LIMITER: ON

↓ ↑ FCC LIMITER: OFF
The FCC Limiter system limits the amplitude of the analog output to meet the requirements for the installation, whether it be 2-wire or 4-wire.

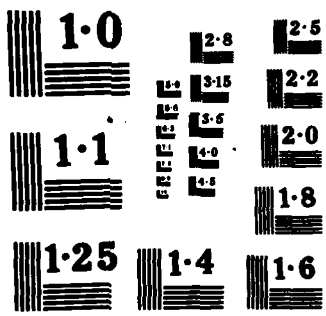
SKIP - INPUT: PBX

↓ ↑ INPUT: HANDSET
All telephone interfaces are included in the PBX input mode. The HANDSET input is used when a handset is plugged directly into the rear of the unit. Audio is still sent to the handset earpiece when PBX input is selected.

AD A156 760 TEST AND EVALUATION OF VIDEO TELECONFERENCING AT 56 KBPS(U) DELTA INFORMATION SYSTEMS INC HORSHAM PA MAR 85 NCS-TIB-85-3 DCA100-83-C-0047 2/2

UNCLASSIFIED F/G 17/2 NL

END
DATE
FILMED
9 85


1·0
2·8
2·5
3·15
2·2
3·5
1·1
4·0
2·0
4·5
1·8
1·25
1·4
1·6

DISPLY - This pushbutton provides for selection of both the alphanumeric display and the bar indicator display.

SKIP - DISPLAY: CALLS - This is the most common operating display. This mode monitors the call status of the line the Vopac is hooked to. The display indicates whether the line is IDLE, has an INCOMING CALL or an OUTGOING CALL. Once a tie-trunk line has been established between Vopac units, calls can be placed through the units to a particular station in the remote PBX. The number dialed is displayed as it is dialed. SYNC LOST is a display that indicates a failure in the data line that is preventing communication with the remote unit. A communication loss could be due to a disconnection in the data link. It may also be caused by having the units not set to the same operating mode.

↓↑ DISPLAY: ECHO RET - This mode is primarily used as an installation/diagnostic aid to monitor the undesired audio signal level returned. The display format is: ECHO RET: XX DB.

↓↑ DISPLAY: ALC GAIN - This mode displays the input gain changes caused by the automatic level control system. The display format is:
FROM: XX TO: XX.

↓↑ DISPLAY: OUTPUT - This mode provides an alphanumeric output level. The display format is: OUTPUT: -XX DBM.

SKIP - BAR: XXX - This function determines what information is displayed on the bar graph on the front of the Vopac.

↓↑ BAR: INPUT LEVEL - This mode monitors the peak input level of the incoming audio data. The display is a linear display. To facilitate peak readings, the peak detector keeps the peak level LED of the bar graph lit for approximately one-half second after each change. This function is useful for setting the input gain levels on the phone line (PBX) input. Normally, voice peaks should be set to read about one-third to one-half scale, with the upper half of the range reserved for headroom. However, the audio input circuits are fully linear up to full scale, and only inputs which "peg" the meter result in actual input overload.

DISPLY (cont'd) ↓↑ BAR: NOISE FLOOR - This is primarily a diagnostic mode used for displaying background/line noise, hum, etc. In this mode the bar graph displays low level input signals that would be too small for the normal INPUT LEVEL display. A quiet input line should read in the left quarter of the display. In the NOISE FLOOR mode, the bar graph displays a true RMS measurement, rather than a waveform peak measurement. Each LED segment represents 1 DB, with the leftmost LED lighting for an input at 65 DB below the full scale input overload.

↓↑ BAR: IN & OUT LEVEL - This mode simultaneously displays the input audio level as a left to right bar graph and the output audio level as a right to left bar graph. Each bar graph has identical characteristics to the input only INPUT LEVEL display.

INFO - This button will cause the current software version number to be displayed.

Chapter 4:

TROUBLESHOOTING

Introduction

The Vopac is a reliable unit and problems in completing a communication link will frequently be caused by improper function settings. *Table 5 lists communication problems that may be* encountered, along with the Vopac function settings that should be checked as a possible cause of these difficulties.

A description of the loopback delayed-voice test is also given. The loopback test is used to determine the functionality of the local unit as well as the integrity of the data link to the remote Vopac. The Self-Test Diagnostics program is used to determine if problems are due to component failure within the unit.

It should be noted that in the event a malfunction is traced to a particular section within the Vopac, the availability of a spare unit will allow service to continue within the communication system while the failed unit is repaired.

Power Up

In case of a failure of the unit to "power up" when the switch on the front panel is turned on, check the following:

1) Line cord: Both ends of the AC line cord should be firmly seated in the back of the Vopac and in the wall outlet.

2) Fuse. Check that the fuse, which is located at the back of the unit, next to the AC line input plug, is not "blown". The fuse can be visually inspected for no break in the wire inside of the fuse, or tested with a ohmmeter. Note: if the fuse is blown proceed directly to point 3) before inserting a new fuse.

3) Input voltage selector: The value of the input voltage that the Vopac is set for appears directly below the fuse. If this is not the correct voltage, remove the selector card and re-insert it with the proper voltage value displayed so that it can be read from the back of the unit. If you have proceded to this step after finding a blowm fuse, and the voltage number is correct, replace the fuse and turn power on. If the fuse again blows, the Vopac has experienced a power supply failure and must be repaired.

4) Wall circuit: Make sure that there is power at the wall outlet.

5) If the unit still fails to power up, consult local service representative.

Delayed Voice Loopback Test

The delayed voice loopback is designed to quickly isolate problems to the local unit, the remote unit or the digital link. When a Vopac unit is determined to be at fault, the built-in diagnostics can be used to help locate a failure within the unit.

The delayed voice loopback test can be used by a single user to test an individual Vopac unit, although a thorough check of the system would be simplified with an operator at each end and a means of communicating between sites. Perform the loopback test on each of the units as follows:

1) Using the CHANL pushbutton, SKIP to LOOPBACK: XXX and turn LOOPBACK: ON with ↓ or ↑ .

2) Plug a test handset into the back of the unit or use a Watson ringdown phone.

3) Speak into the phone. The speech should be heard in the earpiece delayed approximately 300 msec.

In this test, the voice signal is analyzed, digitized, then immediately looped back to the synthesizer and converted back to an analog signal. This demonstrates the proper functioning of the digitizer and synthesizer portions of the unit, as well as a good portion of the analog interface.

The digital link between the two units can be tested by systematically looping the signal through each piece of equipment between the Vopacs, until all devices in the link have been tested. The loopback test for the entire digital link is performed as follows:

1) In the remote Vopac, set LOOPBACK: ON.

2) Insure that the local Vopac is is PBX: RINGDOWN if the Watson phone is used or INPUT: HANDSET if a handset is used.

3) Speak into the phone. The operator's speech should be heard in the earpiece delayed approximately 300 msec.

With the remote Vopac in loopback mode, the incoming bitstream is looped back to the outgoing bit stream. This means that the local Vopac receives as input the same signal it output after travelling the entire data link in both directions.

The same procedure should be followed to test each Vopac and the data link from both ends. After each end is tested, verify total system operation by talking end-to-end using the test handset.

If the Vopac system has passed all tests for the data link and a problem still exists in using the PBX or other communication system, the telephone interface must be checked, including both the voice and signalling interfaces.

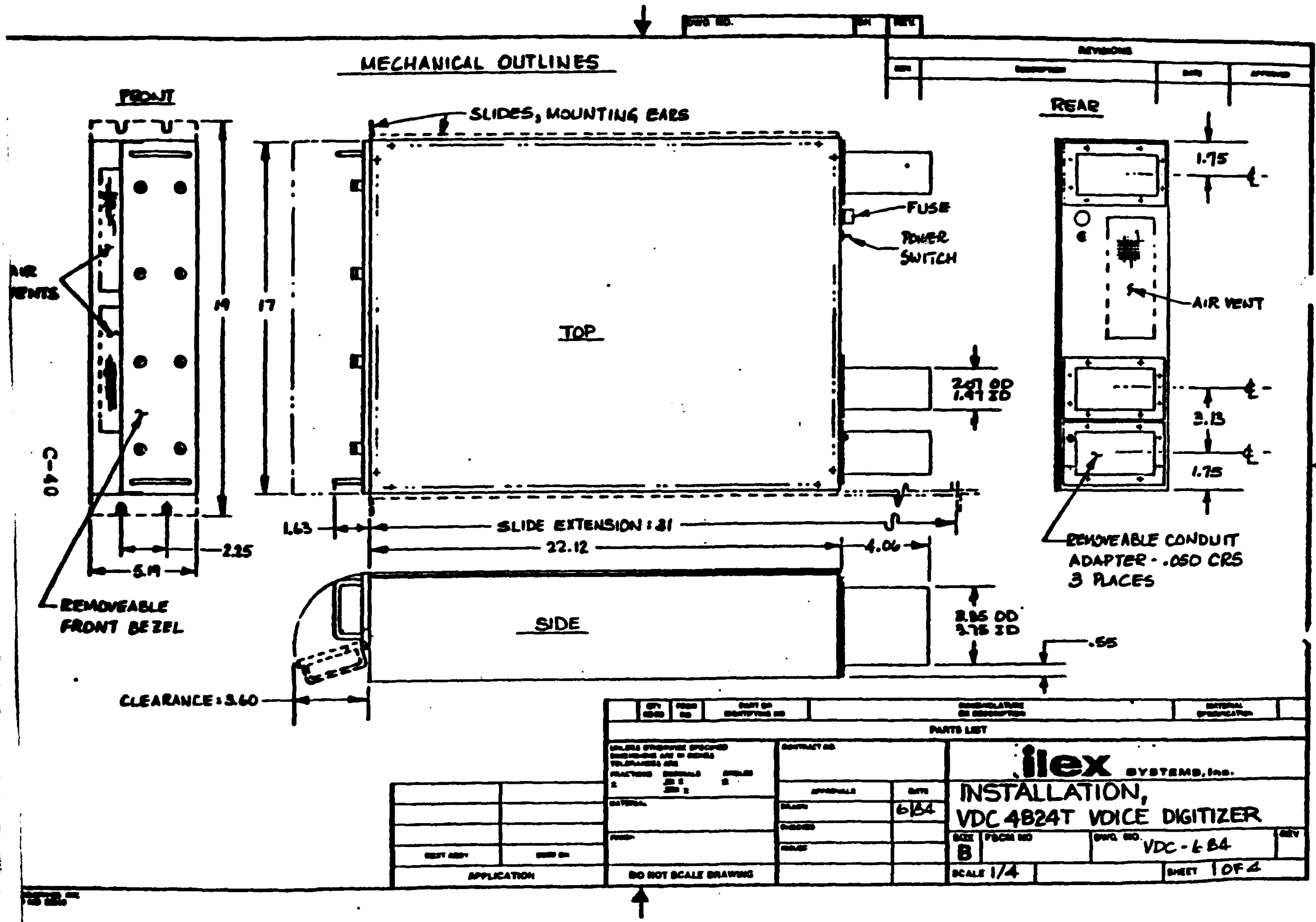
Self Test

The self test can be used to isolate a problem to a particular section within the Vopac when it has been determined that a unit is not operating properly. The self test is initiated by simply depressing the TEST button, pressing SKIP until SELF TEST: OFF appears on the display, and finally pressing ↓ or ↑ to select SELF TEST: ON. The Vopac steps through and individually tests the analog circuitry, the memory circuits, and the high speed numerical processor, giving an indication of PASS or FAIL as each section test is completed. After all sections have been tested, a summary is displayed indicating that all tests have passed, or which board within the unit contains the bad section and needs to be serviced.

In the event a failure is detected, it is recommended that a spare vopac unit be installed so that communication can continue while the failed unit is repaired.

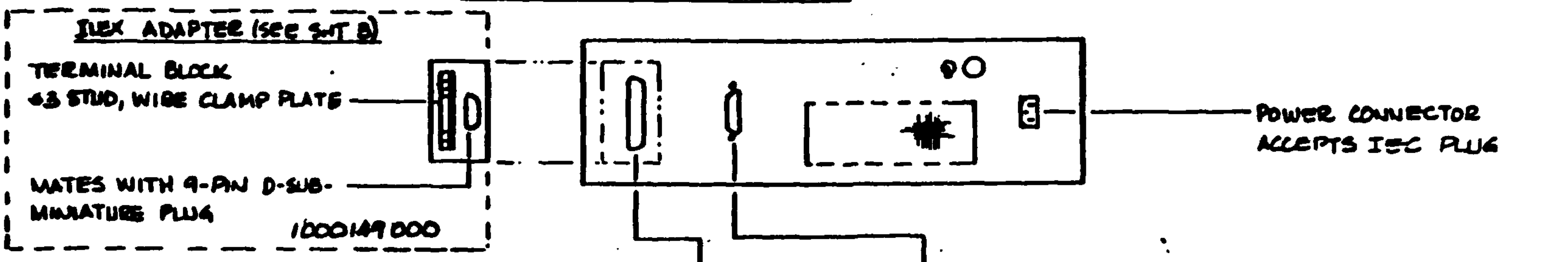
PROBLEM	FUNCTION SOLUTIONS
No voice communication or "SYNC LOST"	BAUD RATE -- check that both units are set to the same baud rate.
	CLOCK -- must be set to EXTERNAL if a mode is used or some other device which provides line clock. Set to INTERNAL otherwise. If only one Vopac is set for wrong clock source, communication will be allowed towards correct unit, but incorrect unit will not work properly.
	LOOPBACK -- loopback must be OFF.
	LPC10/DBSS -- in 2400 baud rate, both units must be set to the same type of algorithm, either LPC10 or DBSS.
	PBX/HANDSET--must be set to HANDSET only for handset use. Otherwise use PBX. Audio will be output to the handset in PBX mode, but input is not allowed by another phone system while the unit is in HANDSET mode.
	RINGDOWN/EXTENSION/TIETRUNK -- check for correct setting for proper operation.
One way communication only	see CLOCK above
Unable to dial through remote PBX	DTMF/ROTARY -- must be set properly for type of dial system connected to.
	10/20 PPS -- must be set to correct rate for rotary system connected to.
Audio problems: Echo	ECHO CANCEL -- set ECHO CANCEL: ON.
Voice too soft	TO PBX: / FROM PBX: -- it is most beneficial to set system voice levels with TO PBX: using the ↓ and ↑ arrow keys. FROM PBX: can be used locally to adjust gain, although it is not optimum.
Breakup or Voice too loud	TO PBX: / FROM PBX: -- these levels should be kept below about 36 db as voice peaks above this level will cause breakups and distortion. If problems persist, try turning ALC GAIN OFF, then ON.

TABLE 5 -- TROUBLESHOOTING COMMUNICATION PROBLEMS



MECHANICAL OUTLINES
FRONT
TOP
SIDE
REAR
SLIDES, MOUNTING EARS
FUSE
POWER SWITCH
AIR VENT
AIR VENTS
REMOVEABLE FRONT BEZEL
REMOVEABLE CONDUIT ADAPTER - .050 CRS 3 PLACES
SLIDE EXTENSION: 21
22.12
4.06
1.63
19
17
2.25
5.19
1.75
3.13
1.75
2.07 OD
1.97 ID
2.85 OD
2.75 ID
.55
CLEARANCE: 3.60
PARTS LIST
ilex SYSTEMS, Inc.
INSTALLATION, VDC 4824T VOICE DIGITIZER
B
VDC-684
SCALE 1/4
SHEET 1 OF 4
6/84
DO NOT SCALE DRAWING
APPLICATION

REAR PANEL CONNECTIONS
(CONDUIT ADAPTER REMOVED)



MATES WITH 50-PIN RIBBON-TYPE PLUG OR ILEX ADAPTER 1000149000.

MATES WITH 25-PIN D-SUBMINIATURE RECEPTACLE RS-232C OR MIL-STD-188C.

TELEPHONY CONNECTOR

PIN NO.	SIGNAL NAME	FUNCTION
1	TT	AUDIO INPUT TO VDC
26	TR	AUDIO INPUT TO VDC
2	RT	AUDIO OUTPUT FROM VDC
27	RR	AUDIO OUTPUT FROM VDC
3	BAL1	BALANCE FOR 2/4 WIRE
28	BAL2	BALANCE FOR 2/4 WIRE
17	EP	E SIGNALLING, POSITIVE NODE
42	EN	E SIGNALLING, NEGATIVE NODE
12	EMR	E SIGNALLING, AUXILIARY
25	RDETP	RING DETECT, POSITIVE NODE
50	RDETN	RING DETECT, NEGATIVE NODE
35	PLOOP	M SIGNALLING, LOW IMPEDANCE
11	MP	M SIGNALLING, POSITIVE NODE
36	MN	M SIGNALLING, NEGATIVE NODE
19,44	BATTGND	VDC INTERNAL 48VDC SUPPLY
20,45	BATT	VDC INTERNAL 48 VDC SUPPLY
9,34	DG	SIGNAL LOGIC GROUND
13,38	+5VDC	+5VDC, 100 OHM SOURCE
14,39	+15VDC	+15VDC, 100 OHM SOURCE
15,40	-15VDC	-15VDC, 100 OHM SOURCE

DATA CONNECTOR

PIN NO.	SIGNAL NAME	FUNCTION
1	FG	FRAME GROUND
2	TXD	TRANSMITTED DATA
3	RXD	RECEIVED DATA
4	RTS	REQUEST TO SEND
5	CTS	CLEAR TO SEND
7	SG	SIGNAL GROUND
8	DCD	DATA CARRIER DETECT
15	TXC	TRANSMIT CLOCK
17	RXC	RECEIVE CLOCK
20	DTR	DATA TERMINAL READY
24	ETXC	TRANSMIT CLOCK GENERATED BY VDC

PARTS LIST

ilex SYSTEMS, Inc.

INSTALLATION, VDC 4824T VOICE DIGITIZER

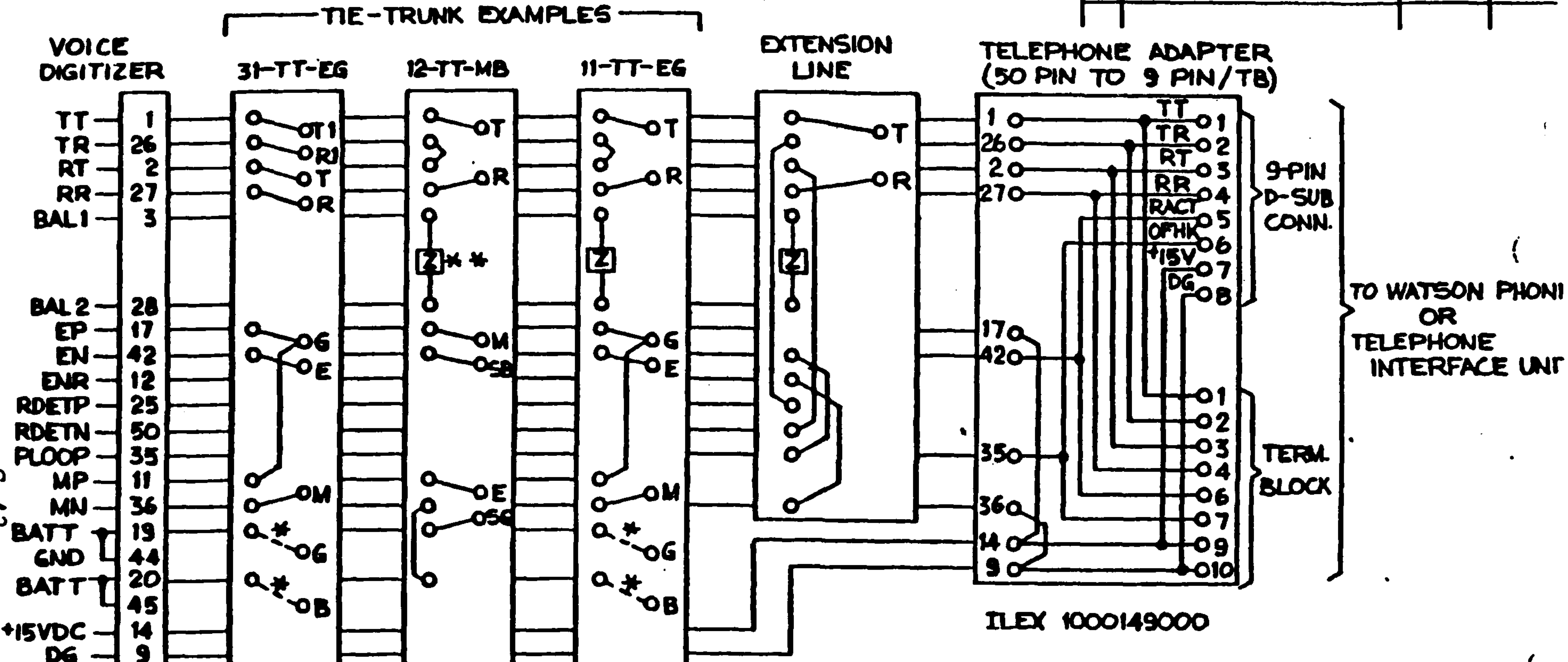
SIZE B | DWG. NO. VDC-684 | SCALE — | SHEET 2 OF 4

APPLICATION

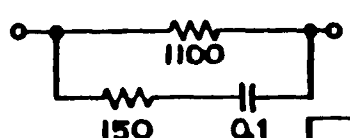
DO NOT SCALE DRAWING

C-41

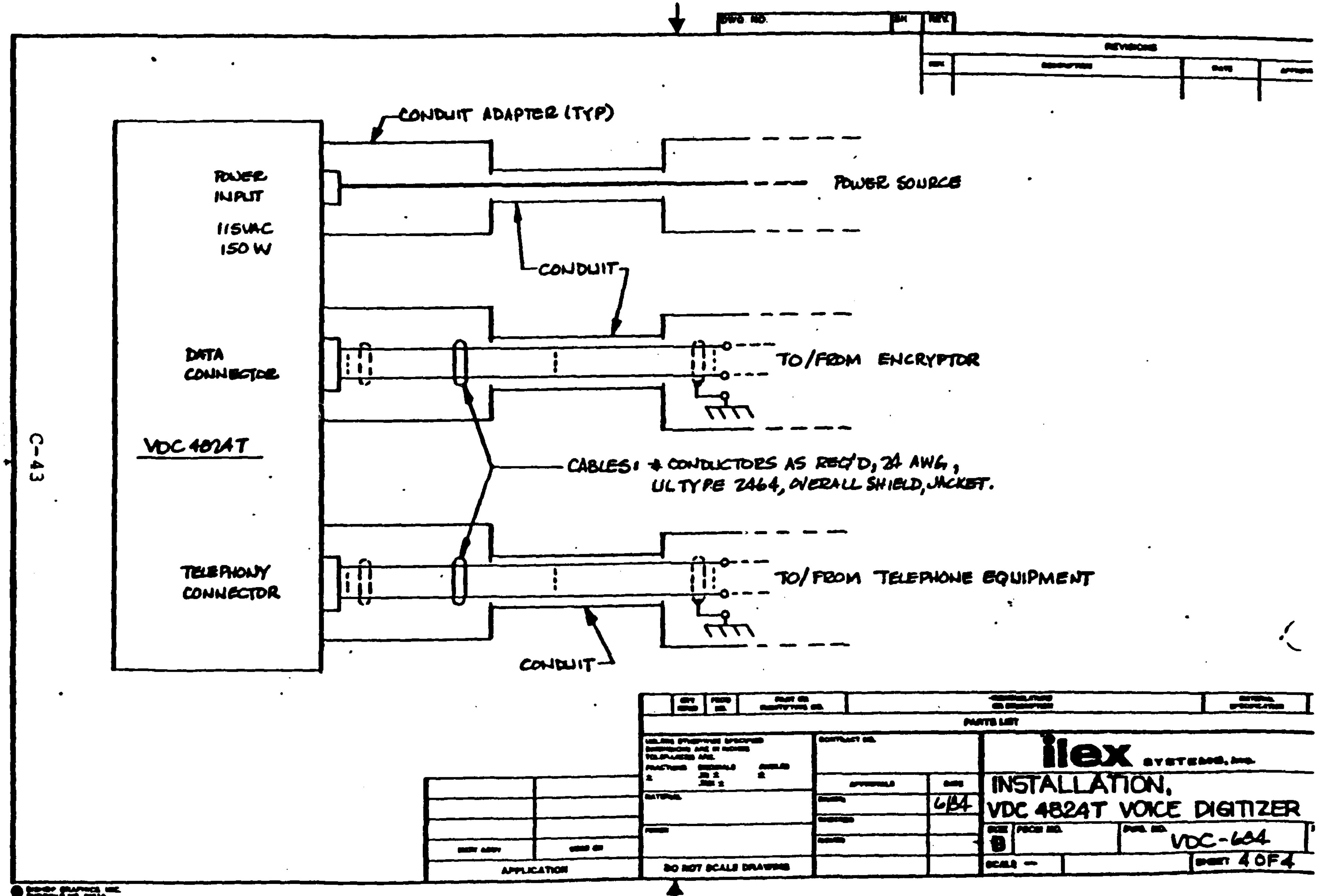
TELEPHONY CONNECTOR OPTIONS



* OPTIONAL CONNECTIONS REQUIRED IF POWER IS NOT AVAILABLE FROM TIE-TRUNK.



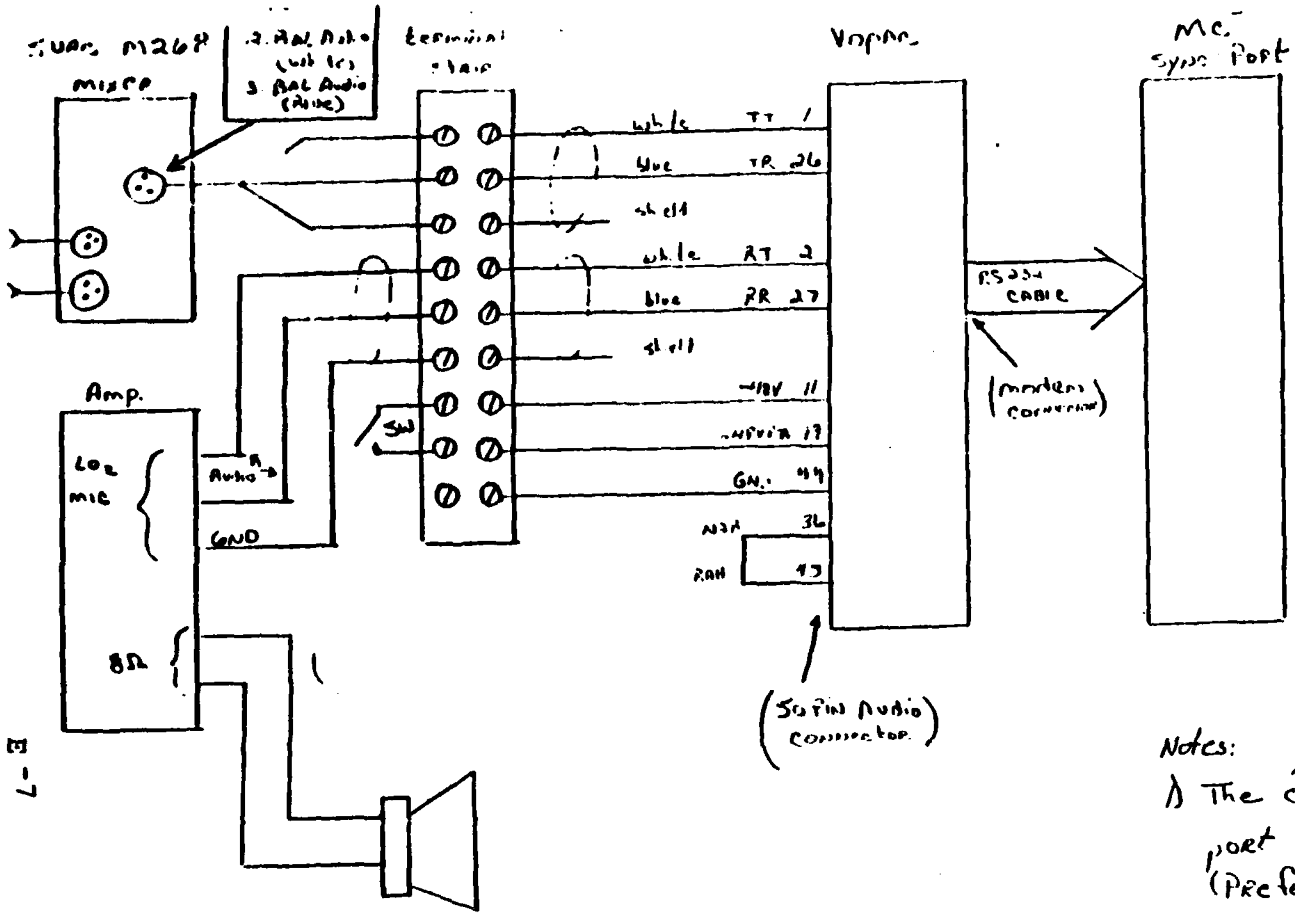
PARTS LIST	
ilex SYSTEMS, Inc.	
INSTALLATION, VDC 4824T VOICE DIGITIZER	
DATE	6/84
SIZE	B
DWG. NO.	VDC-684
SHEET	3 OF 4
	DO NOT SCALE DRAWING
	APPLICATION


CONDUIT ADAPTER (TYP)
POWER INPUT
115VAC
150 W
POWER SOURCE
CONDUIT
DATA CONNECTOR
TO/FROM ENCRYPTOR
VDC 4824T
CABLES: # CONDUCTORS AS REQ'D, 24 AWG, UL TYPE 2464, OVERALL SHIELD, JACKET.
TELEPHONY CONNECTOR
TO/FROM TELEPHONE EQUIPMENT
CONDUIT
PARTS LIST
REVISIONS
APPLICATION
DO NOT SCALE DRAWING
ilex SYSTEMS, INC.
INSTALLATION,
VDC 4824T VOICE DIGITIZER
6/84
B
VDC-684
4 OF 4

APPENDIX D

VIDEOTAPE TRANSCRIPT

VOPAC to MC



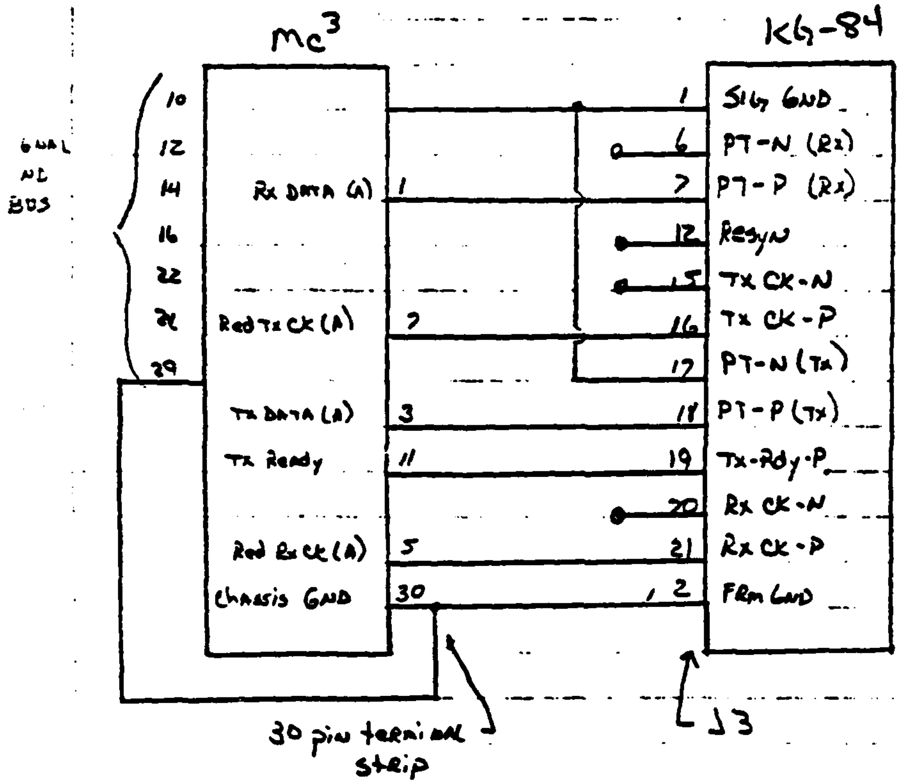
Notes:

1) The 'control rate' for the vopac sync port must be programmed to '1' or higher (Preferably '1'). Programming the CR for 0 results in the port carrier detect line going low and pulsing high when the system recycles, this forces the Rx data to be clamped to all 1's. (effectively the Rx side of the ckt will be dead). A CR of 1 or higher results in normal operation of carrier detect (high - then low as system resyncs), with the CR set to 1 or higher the vopac momentarily goes dead as the system resyncs - then returns to normal operation.

F. KG-84 to DSU (Black side) MASTER

DSU		Wire		KG-84
SIG GND	B	GREY	1	SIG GND
Rx DATA (A)	R	ORANGE	12	RCT-P
Rx DATA (B)	T	PURPIE	13	RCT-N
Rx ck (A)	V	GREY	19	RCTC-P
Rx ck (B)	X	Red	20	RCTC-N
Tx DATA (A)	P	Blue	14	TCT-P
Tx DATA (B)	S	BROWN	15	TCT-N
Tx ck (A)	AA	Black	21	BSC-P
Tx ck (B)	Y	white	22	BSC-N
Ex ck (A)	U	Green		
Ex ck (B)	W	yellow		

E. Mc3 to KG (Red side)

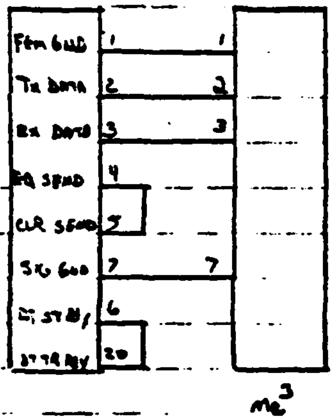


B. VOPAC 4800 baud

The VOPAC is supplied with an RS-232 interfacing cable. Connection to ME^3 is made by connecting VOPAC Modem jack to ME^3 port 5 via RS-232 connector. No other wiring is necessary for this interface.

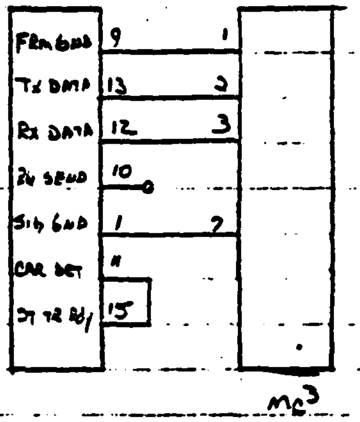
C. Silent 700 (order wire) 300 baud

NOTE: Run TI's in half dux/local copy mode for order wire.



Silent 700 (25 pin)

Port 6



Silent 700 (15 pin)

port 6

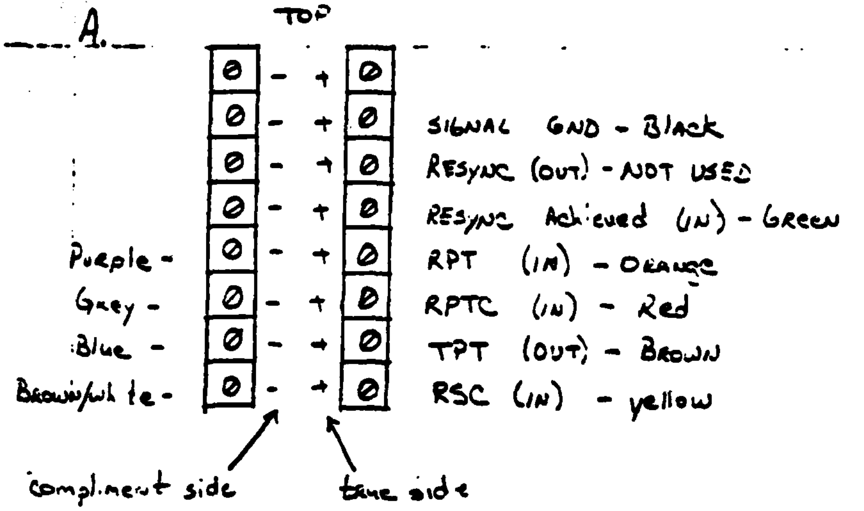
NOTE: Cable may be moved to control port without any modification.

D. Control terminal (300 baud)

Same as C.

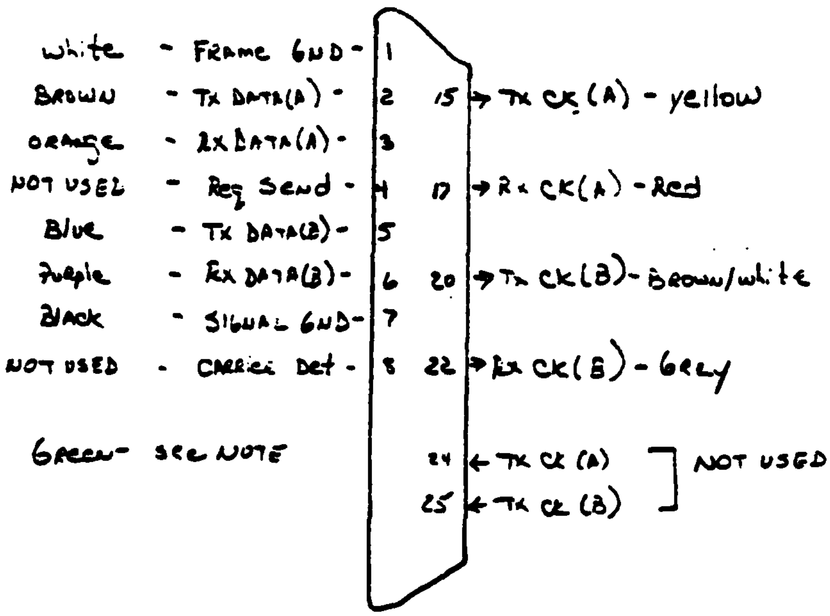
NOTE: TI's must be set for capital letters.

A.

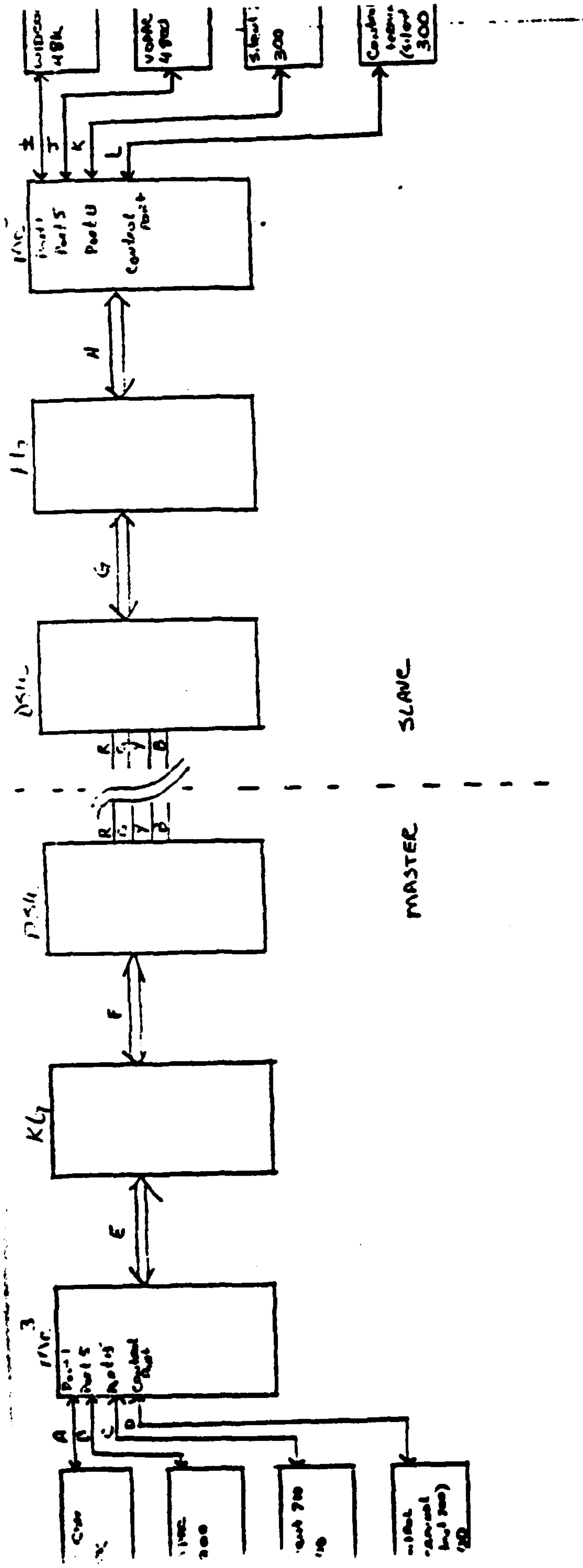


Note: Be sure to count terminals from the bottom of the interface. Both units do NOT have the same number of terminals.

NOTE: GREEN WIRE is brought out of Port connector AND tied to (pin 19 Resync Achieved) GREEN wire from Red K6 connector At Aggregate terminal strip pin 27. Pin 27 is not used except as a junction for the resync Achieved Lin

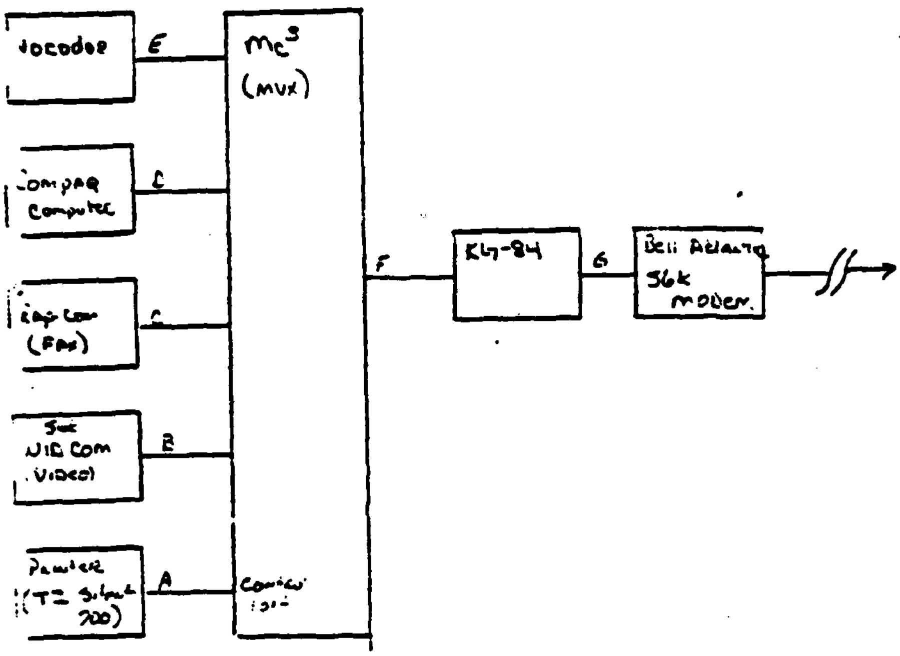


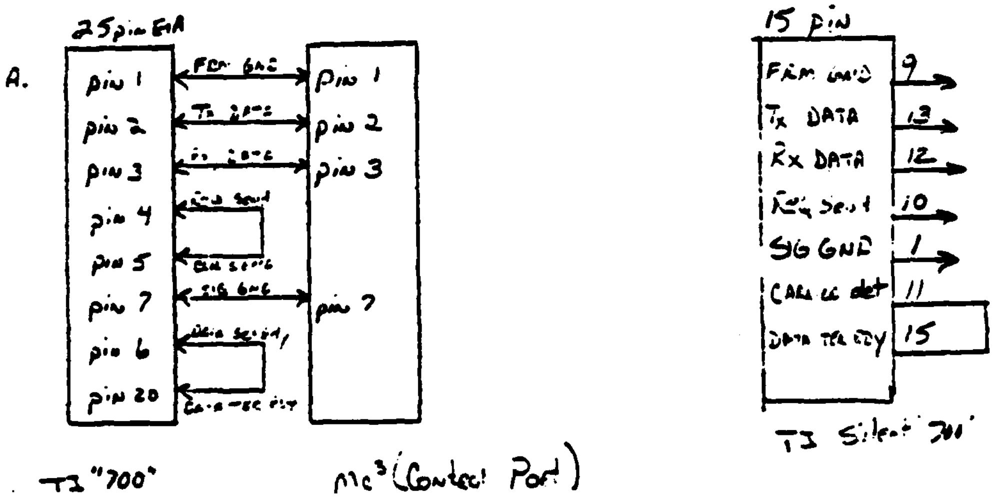
Port 1 of mc^3



MASTER
SLAVE
KL7
A
B
C
D
E
F
G
H
K
L
R
Y
D
300

System Block Diagram





Control terminal Connections.

NOTE: These same connection apply for an order wire channel. (sync - 300 baud). place old TI in half dux - new TI in local copy for operator copy at the tx end.

APPENDIX E

DOCUMENTATION OF TELECONFERENCING SYSTEM OPERATING AT 56 KBPS

hour, two-way videoconference, coast to coast, for about $60.

We'd like your opinion. After you've seen the system in operation, please fill out one of these short evaluation forms and leave it with the coordinator at your site. If you would like more technical details on this system or on video compression, there are block diagrams available and copies of this recent Codec Comparison study, developed under contract to the National Communications System.

Enjoy the demonstration!

times the price of a phone call.

Even more important, AT&T plans to make that network available in more than one hundred cities within two or three years. There's one other way this system has developed. It's gotten smaller. Where earlier Codecs weighed hundreds of pounds, the particular Codec used in this test weighs only 65.

It's also important to note that digital signals are much easier to encrypt than analog signals. So, like other video compression Codecs, the digital output of this equipment is capable of being fully encrypted, supporting the highest level of security.

Of course, this system represents just one combination of communications equipment that is now on the market. The microphones, cameras and monitors used are standard and can be changed to suit individual requirements. But the real goal of this test is to get your evaluation of the benefits and drawbacks to this type of videoconferencing.

It is worth giving up some picture quality to get:

Nationwide accessibility - the ability to reach most major cities, and to utilize existing communication links that agencies already have at their disposal,

Portability - the ability to move the equipment and use it FROM many different locations, as needs arise,

Low Cost signal transmission - the ability to transmit a one

At each of these stages of development, the manufacturers of the compression systems, the CODECS, tried to maintain good picture quality at these lower speeds. But even these reduced band-widths are high enough to require special transmission lines.

And they're still too bulky to be handled conveniently through a switched network, to provide videoconferencing connections that are as simple as placing a phone call.

In order to make videoconferencing as accessible and as inexpensive as possible, it was necessary to compress that signal another ten-fold, to 56 Kbps. That's like going from this size bandwidth to SOMETHING much smaller than this.

AT&T will soon be offering 56 Kb switched digital service between major cities throughout the country, and the phone wires already in place, in agency offices for example, will be able to be converted to this service.

But this last ten-fold compression of the video signal made it difficult to maintain the image quality. As you'll see, the picture is fine until motion is introduced or until the image is changed. Then the Codec requires a little time to "catch up" with the changes. That's not surprising, since the original signal bit rate was almost 1500 times larger than the bit rate used by this system.

So in giving up some picture quality, what's been gained? This 56 Kb network will offer low cost transmission, for one thing. Coast to coast videoconferences will cost about 2 1/2

each new video frame in a series only has to be updated, the transmission bandwidth requirements are reduced.

Cost savings with video compression are... impressive. The picture you're watching is standard analog video. In digital terms, it's transmitted at 80 million bits per second. For interactive, two-way videoconferencing, it would take two of those circuits. The equipment necessary to send these signals just fifty miles could cost half a million dollars, and the use of that fifty mile circuit could cost an additional 15 to $20,000 per month. The compressed video images you'll see in this test can be sent coast to coast, in both directions, for less than $100 per hour.

To illustrate what's happened so far, we'll use this picture as an analogy. Analog video, or straight digital video without any compression, requires 80 Mbps for transmission. We'll say that's the entire video screen. As video compression technology developed, bit rate requirements dropped from 80 Mbps to 36, about THIS MUCH of a reduction. Then they dropped to 6 Mbps, about HERE. At that point, videoconferencing began to look economically possible. But the technology continued to develop and the bit rate requirement was cut in half AGAIN. And AGAIN, to 1 1/2 Mbps. This is the transmission rate used right now by AT&T for their Picturephone Meeting Service. Then, during the past year, systems have been installed that run very effectively at 768 and even 512 Kbps. That's less than *one percent* of the original signal requirement, and each of these reductions has meant dollar savings to users.

A few years ago, it became possible to translate video signals from ANALOG to DIGITAL format. This by itself didn't solve the problems for videoconferencing because the digital version of the analog signals still required a wide band-width. But Digital signals can be manipulated in many ways, and it's this ability to manipulate that led to the concept of video compression.

In straight ANALOG or DIGITAL video signals, the picture is broken down into thousands of PIXELS, and each of those picture elements is translated into an electronic message and sent out. But within any single video frame, and even more, from one video frame to the next, most of those picture elements are the same. In the picture you're watching, for example, the background is fairly uniform. So if the picture is broken down into sections, like this...

quite a few of those sections are duplicates. Especially in the background, there is no change from one block to the next. Instead of sending all the information on each of those blocks, digital signals make it possible to send that information just once, for the first section, and then send a brief code that says, effectively, do it again, do it again.

There's even more duplication from one video frame to the next. If we freeze my image...

...and compare two frames, you can see that most of the picture stays the same. Video compression takes advantage of this fact and sends only the changes, rather than the entire image. Since

APPENDIX D

VIDEOTAPE TRANSCRIPT

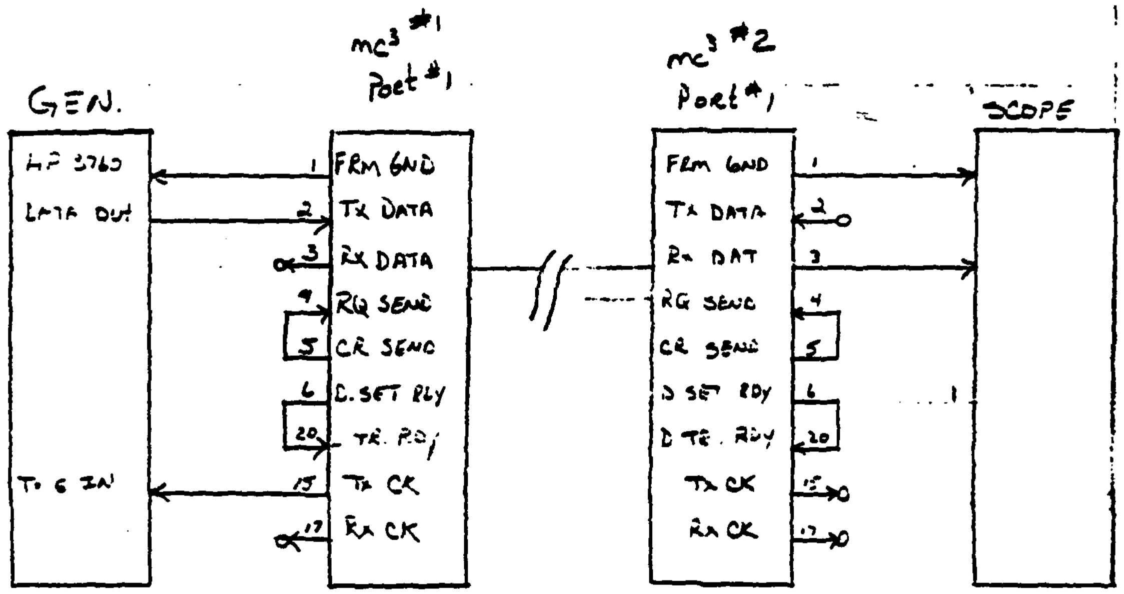
On behalf of the National Communications System and the Federal Emergency Management Agency, I'd like to welcome you to this videoconferencing evaluation experience and give you a little background on the communication system you'll be using.

I'm Bob Keiper, with Delta Information Systems. As consultants in teleconferencing, we've seen great improvements in this technology during the past few years - and the equipment assembled for this test represents the leading edge of this developmental cycle.

It isn't broadcast television. You'll see some blurring of the picture when people move, and you'll notice that, although graphic images are clear, they can take a little time to form. In other words, it might not look like leading edge technology. Let me show you why we think it is.

In order for videoconferencing to be practical, it needs to be relatively inexpensive, and it needs to provide point to point connections between many sites, like the "switched networks" of the phone companies.

But standard television signals are broadcast in what's called ANALOG format. They require a lot of bandwidth, which translates into high cost. Fine for broadcast television, but much too expensive for "electronic meetings".

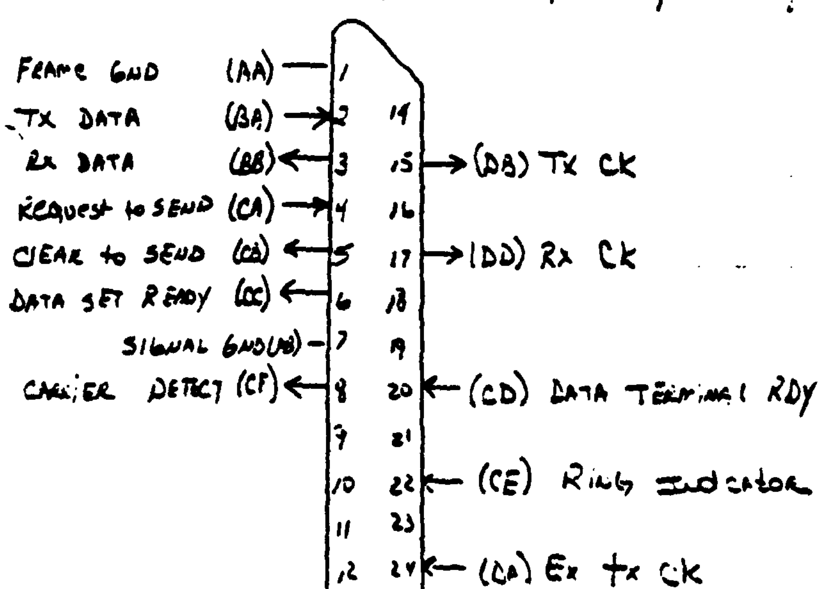


1. Mc^3#1 - Port#1 is configured here as Tx - Mc^3#2 - Port#1 As Rx.

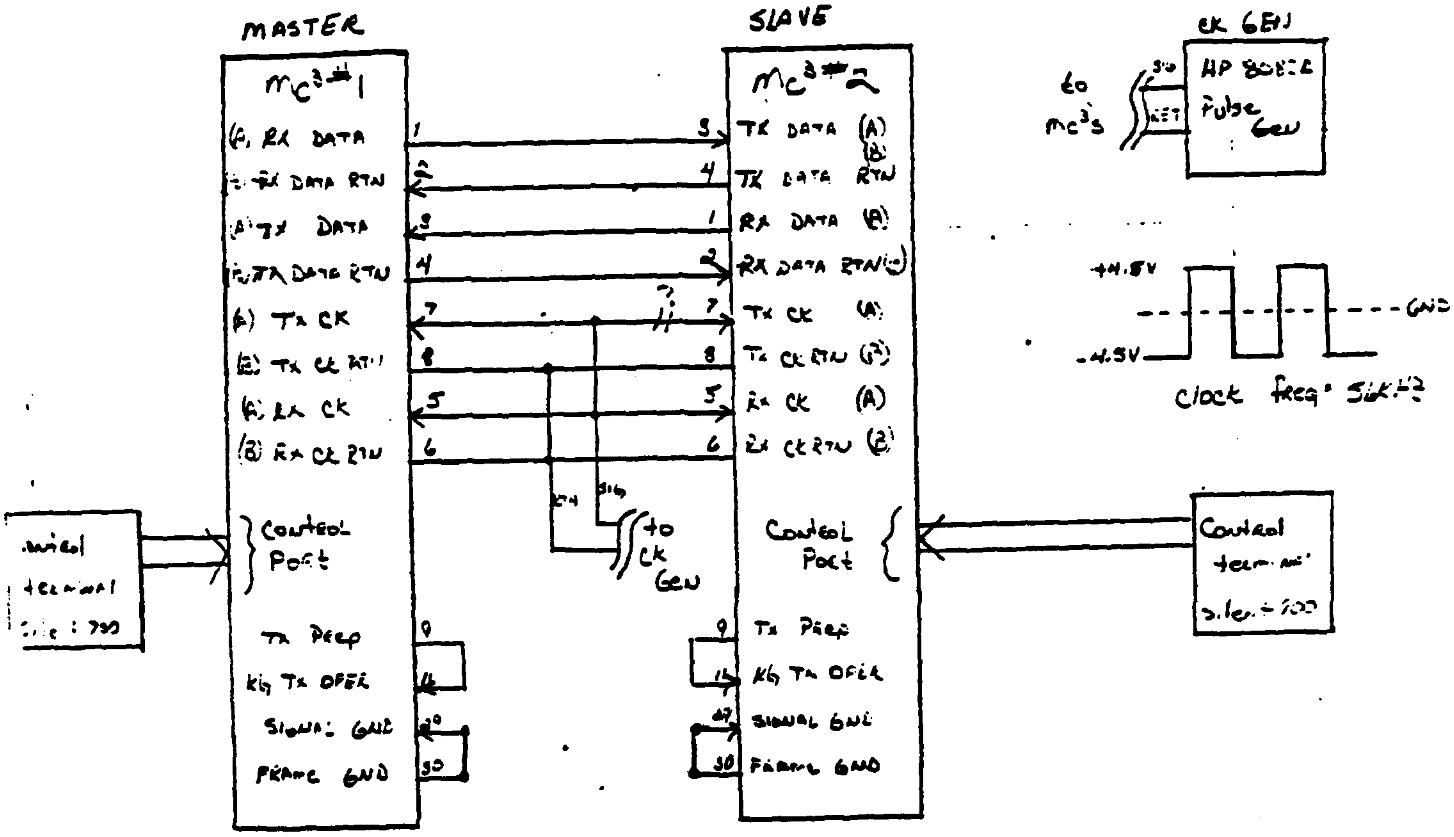
2. Mc^3#2 - Port#1 Rx data was looped to Tx data - Rx data was then monitored on Mc^3#1 - Port#1 Rx side.

3. Remember: Frame GND must be tied to Signal GND (pin 29 to pin 30) of Mc^3 terminal strip (Aggregate side) before data will pass through mux. LE loops were S/S, but LI were inoperable without gnd strap.

4. Our machines have 'D' version sy. cards - port pin assigns as follows:



Mc^3 to Mc^3



Notes: 1. ALL Mc^3 connections made to 30 pin terminal strip (located in back of unit) - Aggregate (trunk) side of Mc^3's.

2. Mc^3#1 programed as follows:

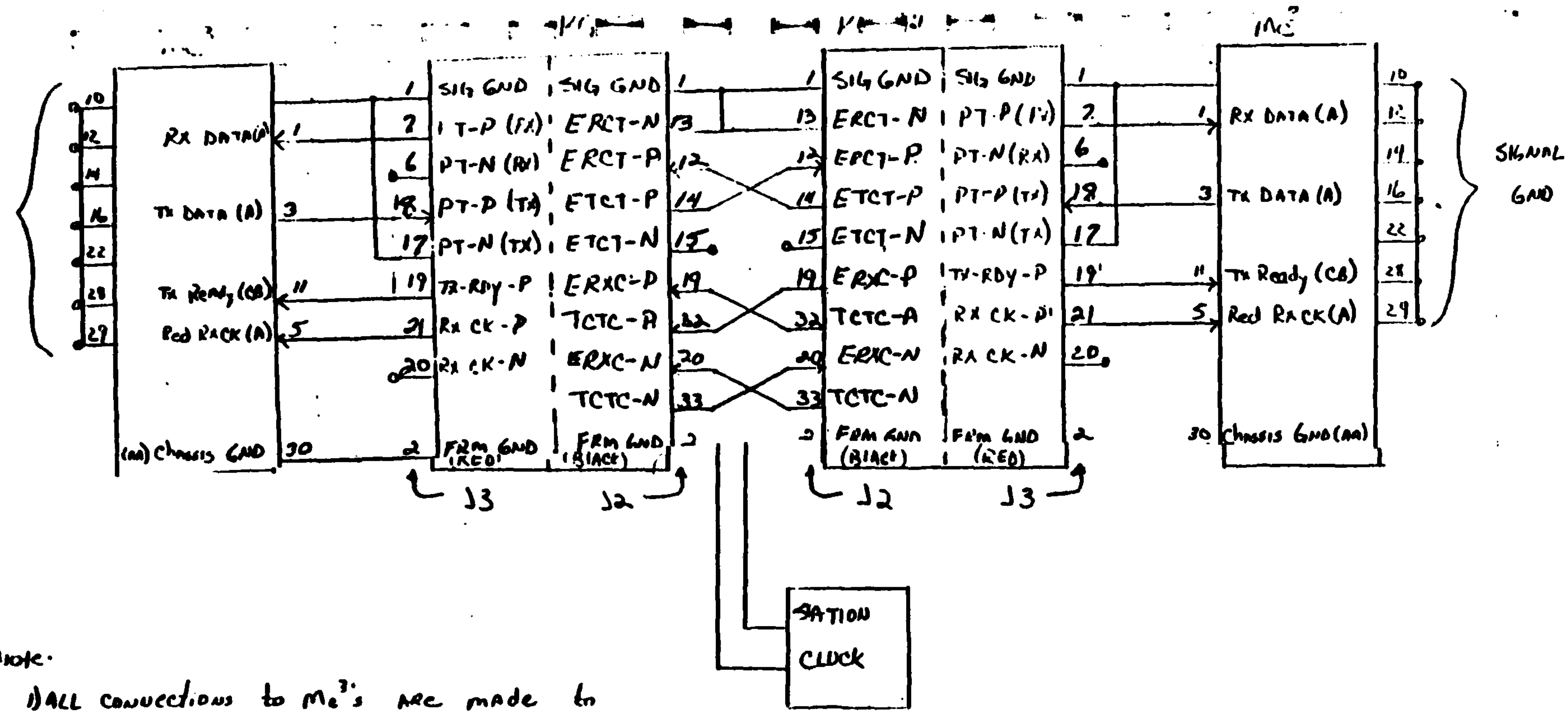
EA (examine active)
TC * RC R*56000 LB*N
P1 TY*SY R*2400 LB*N C*IN CR*0
P2 TY*SY R*2400 LB*N C*IN CR*0
P3 - P16 spare

SV LD*10 SA*10 PIS*N MS*M TC*N SATM*N SATS*N ES*N

3. Mc^3#2 programed same as #1 except is slave. - MS*S

4. Units gave minor alarm with P1, P2 FLT errors when external clock source exceeded 56kHz.

5. Without pin 9 to pin 11 - units continuously recycle with



Note·

1) ALL connections to Mc³'s are made to Aggregate side - 30 pin terminal strip

2) pin 29 is jumpered to pin 30 of Mc³'s.

3. KG-84 Red Side - Unballanced

Black side - Unbalanced

DATE
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-8